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DIGITAL STANDARDS CONVERSION: interpolation theory and aperture synthesis

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Summary

A frequency domain approach is used to investigate digital interpolation for standards conversion. To illustrate the background theory, one-dimensional sampling and interpolation is described in general terms, including methods for synthesising interpolation apertures and the effects of aperture quantisation. With the two-dimensional vertical-temporal sampling of television scanning, interpolation can be carried out as two one-dimensional processes or, with improved performance but somewhat greater complexity, as a single two-dimensional process. Standards conversion impairments are shown to arise from the interpolation frequency characteristic departing from the ideal low-pass form in particular regions of the television signal spectrum. Also identified are the positions in the vertical-temporal spectrum of cross-luminance, cross-colour and other residual impairments left by PAL or NTSC decoding.

Using an experimental converter, apertures of duration two, three and four field periods have been developed. For two-field apertures the performance is a compromise between movement judder, line flicker and vertical resolution; four-field apertures give stationary pictures of good resolution, free from flicker and other aliasing effects and reduce impairments left by colour decoding. Movement impairments, although significantly less objectionable than with previous converters, remain because of the overlap of spectral components which results with normal television scanning.

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1. Introduction

1.1. Standards conversion

In Europe, the main scanning standard for broadcast television signals uses 625 lines per picture and 50 fields per second. North America and Japan, however, use 525 lines per picture and approximately 60 fields per second. Therefore, a conversion process is needed to allow programmes originating on one standard to be broadcast in countries using the other standard.

The process of scanning used to create a television signal is a sampling process. It converts the three-dimensional image falling onto the camera tube into a one-dimensional signal, that is, a signal with one degree of freedom; the brightness of the image varies with horizontal and vertical position, and with time, but the scanning process produces a single function of time to represent all three parameters. Scanning with a different number of lines per picture or of fields per second amounts to sampling at a different rate. So, standards conversion is merely a process of sampling rate conversion.

With colour television signals, it is necessary first to decode from composite PAL, NTSC or SECAM into luminance and colour difference signals. In this form, the picture information can be transferred from the lines and fields of the input standard to those of the output standard. Suitable picture information for each line of the output standard is obtained by interpolating between the brightness values of the surrounding input standard lines and fields. Finally, the converted signals are recoded to composite form.

1.2. Interpolation

As the central process of a standards converter, interpolation determines the overall performance. Simple methods of interpolation are inadequate and lead to a wide range of different impairments to picture quality, particularly to moving objects and to areas containing vertical detail.

The process consists of the addition of weighted proportions from nearby lines, with the weighting factors being selected according to the relationship in time and position between the input lines and each output line. Thus, it is necessary to store the input signal in order to allow several lines, possibly from different fields, to be made available simultaneously. This combination of arithmetic and storage makes the process well suited to digital implementation. With this form of instrumentation, the performance characteristics predicted from theory for a particular interpolation algorithm can be accurately reproduced in a practical converter.

In previous converters operating between the 525/60 and 625/50 standards, interpolation has been considered primarily in terms of its time domain performance[1,2,3,4]. This is because interpolator action, particularly for movement, is more easily visualised as a series of overlapping images. However, a greater insight into the probable effectiveness of an interpolation method can be gained by considering its frequency domain performance by methods originally described[5,6] for line-rate conversion from the 625/50 standard to the 405/50 CCIR System A standard. This Report describes how these frequency domain methods have been applied to the field-rate conversion process.

In subsequent Sections, the theory of sampling and sample rate conversion is described, first for one-dimensional interpolation in general terms and then in the particular two-dimensional form required for television signals. The occurrence of various types of picture impairment is related to individual features of interpolator performance in the frequency domain and the requirements for improved performance are identified. Also the beneficial action of the interpolator in suppressing cross-effects left by the colour decoding process is explained. Taking account of these factors, practical interpolation methods are derived for both 525/60 to 625/50 and 625/50 to 525/60 conversions and for separate interpolation of the colour signal components.

1.3. Experimental converter

Although the proportions of different impair-

ments present with each interpolation method can be predicted accurately from theory, some impairments are more damaging in their appearance than others. Therefore, a versatile experimental converter[7] was constructed to allow the subjective effect of different impairments to be assessed. In this converter the interpolation method was controlled by a function stored in a read-only memory. By storing several such functions, the performance of different interpolation methods could be rapidly compared, either by switching between methods or by a split-screen comparison of pairs of methods. Thus, it was possible to relate subjective impairments to regions of the frequency characteristic and so to optimise the interpolation by a combination of theory and practical observations.

The experimental converter also incorporated digital decoders with a variety of switched comb filtering options. Thus, it was possible to select the most suitable decoding methods for standards conversion and to match the interpolation methods for the luminance and colour difference signals to best suppress the cross-effects left by the decoder. The effectiveness of different methods of colour decoding is described in a separate report.[8]

Interpolation methods optimised using the experimental converter have been applied in the ACE production converters,[9] manufactured by BBC Engineering Designs Department.

2. Interpolation theory

2.1. Signal sampling and reconstitution

Signal sampling consists of multiplying a continuous waveform, for example, Fig. 1 (a), by a regular series of impulses spaced by the sampling interval T_s, as shown in Fig. 1 (b). This produces the series of weighted impulses representing the sample values, Fig. 1 (c).

Although signal sampling is essentially a time domain process, its effects can also be viewed in the frequency domain. Fig. 2 shows examples of frequency spectra corresponding to the three types of signal waveform shown in Fig. 1. Multiplication of waveforms in the time domain is equivalent to the convolution of spectra in the frequency domain; so when the baseband spectrum, Fig. 2 (a), is convolved with the spectrum of the sampling impulses, Fig. 2 (b), this repeats the baseband spectrum around harmonics of the sampling frequency $f_s(=1/T_s)$, as shown in Fig. 2 (c).

Samples can be returned to the form of a continuous signal in the time domain by low-pass filtering. This consists of convolving the series of impulses with the impulse response of the low-pass filter. In Fig. 3 the low-pass filter impulse response is triangular and each sample is replaced

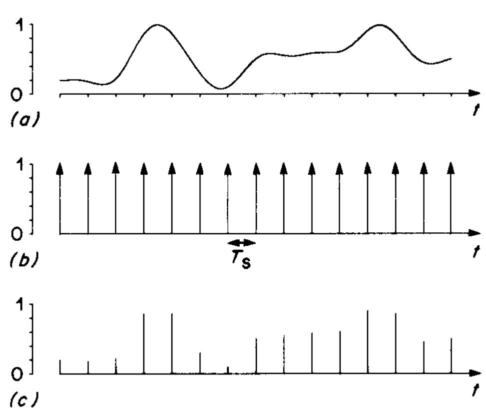


Fig. 1 – Signal sampling: (a) a continuous signal waveform, (b) a series of impulses spaced by the sampling interval T_s, and (c) the weighted impulses representing the sample values, obtained by multiplying (a) and (b).

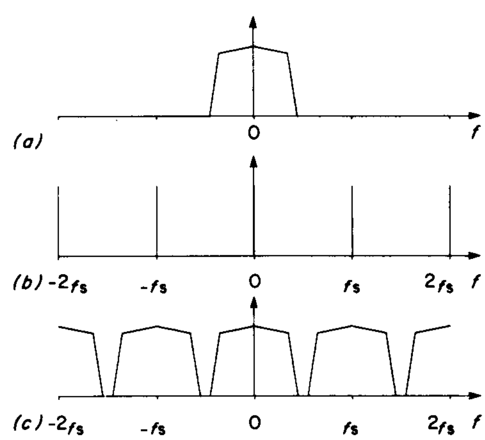


Fig. 2 – Frequency spectra corresponding to the three types of signal shown in Fig. 1: (a) the spectrum of a continuous signal, (b) the spectrum of a series of sampling impulses spaced by $T_s = 1/f_s$, and (c) the spectrum of the sampled signal, obtained by convolving (a) and (b).

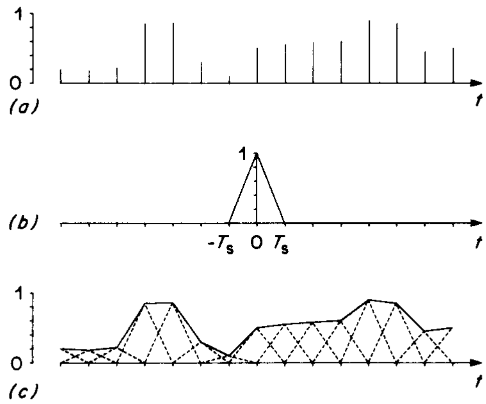


Fig. 3 – Signal reconstitution: (a) the sampled signal of Fig. 1 (c), (b) an example of a low-pass filter impulse response and (c) a continuous signal reconstituted by passing the samples of (a) through the filter (b). Each sample is replaced by the filter impulse response waveform scaled according to the weight of the sample (shown dashed) and these individual responses, when summed, produce the continuous waveform (shown as a solid line)

by this triangular waveform scaled according to the weight of the sample. In Fig. 3 (c), the individual responses, shown dashed, produce, when summed, the approximation to the original waveform shown as a solid line. Although this filter does not accurately reproduce the original waveform, it is not clear how the impulse response should be altered to make an improvement. A much clearer understanding of the action of the filter can be obtained by considering the sample reconstitution process in the frequency domain.

In the frequency domain, the equivalent process to convolution in the time domain is multiplication. Thus, the repeated spectrum of the sampled waveform is multiplied by the frequency characteristic of the filter with the intention of suppressing the harmonic components and retaining only the baseband spectrum. However, the frequency characteristic corresponding to a triangular impulse response is shown in Fig. 4 (b). It is apparent from Fig. 4 (c) that the poor approximation to the original waveform is the result of inadequate suppression of the harmonic spectra, combined with some attenuation of wanted frequencies in the baseband spectrum.

To reproduce all frequencies in the baseband spectrum accurately up to half the sampling frequency would require a filter with the rectangular characteristic shown in Fig. 5 (a). With this 'ideal' filter, the baseband spectrum would be unattenuated and the harmonic spectra completely

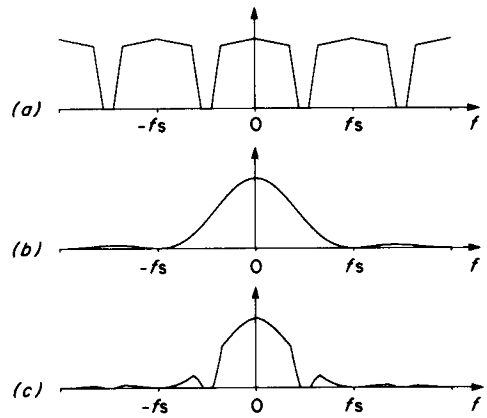


Fig. 4 – Spectra corresponding to the types of signal shown in Fig. 3: (a) the spectrum of a sampled signal, (b) the frequency characteristic of a filter with the impulse response shown in Fig. 3(b), and (c) the spectrum of the reconstituted waveform, obtained by multiplying (a) and (b).

suppressed, thus returning exactly to the spectrum of Fig. 2 (a). However, a rectangular filter has an impulse response defined by

$$g(t) = \frac{\sin \pi f_s t}{\pi f_s t}$$

This $\sin(\pi f_s t)/(\pi f_s t)$ or 'sinc' function, shown in Fig. 5 (b), extends from $-\infty$ to $+\infty$. Therefore, in practice, it is normal to sample at a rate slightly above twice the highest frequency wanted. This leaves a margin for a practicable rate of cut in the reconstituting filter response and so ensures a finite impulse response.

This suggests that one method of improving the accuracy of the reconstituted waveform would be to use a filter with an impulse response which is a better approximation to Fig. 5 (b). For example,

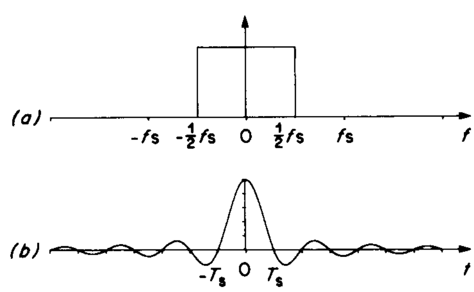


Fig. 5 – An 'ideal' low-pass filter for converting signal samples to a continuous waveform: (a) the rectangular frequency characteristic and (b) the sinc function impulse response.

truncating the sinc function and using a raised-cosine smoothing term so that the impulse response is defined by:

$$g(t) = \tfrac{1}{2}(1 + \cos \tfrac{1}{2}\pi f_s t) \cdot \frac{\sin \pi f_s t}{\pi f_s t} \text{ for } -2T_s \leqslant t \leqslant 2T_s$$

$$= 0 \qquad \text{otherwise,}$$

improves the reconstituted waveform as shown in Fig. 6. In the frequency domain, Fig. 7, the corresponding filter characteristic causes less attenuation of the baseband spectrum and provides better suppression of most harmonic spectra than is given by the triangular impulse response filter (Fig. 4); however, frequencies between $\tfrac{1}{2}f_s$ and f_s are not adequately suppressed. It should be noted that although the impulse responses in Figs. 3(b), 5(b) and 6(b) all have zero values at adjacent sample points, it is not necessary that this should be so to give a good approximation to the original waveform.

If a waveform contains frequencies above the half sampling frequency, then the information content of the samples is irretrievably distorted. Such a situation is shown in Fig. 8, in which the dashed line of Fig. 8(c) represents the waveform obtained by reconstruction with an ideal rectangular filter. Clearly, much of the high frequency content of Fig. 8(a) has been replaced by spurious information of a lower frequency. Fig. 9 shows the

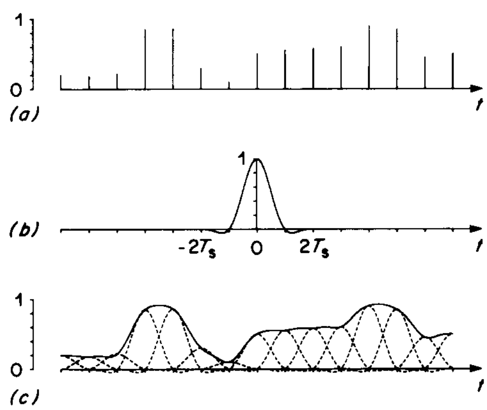


Fig. 6 – Signal reconstitution with a smoothed sinc function filter: (a) the sampled signal of Fig. 1(c), (b) the smoothed sinc function impulse response and (c) the reconstituted waveform (shown as a solid line). The individual impulse responses used to make up the continuous signal are shown dashed.

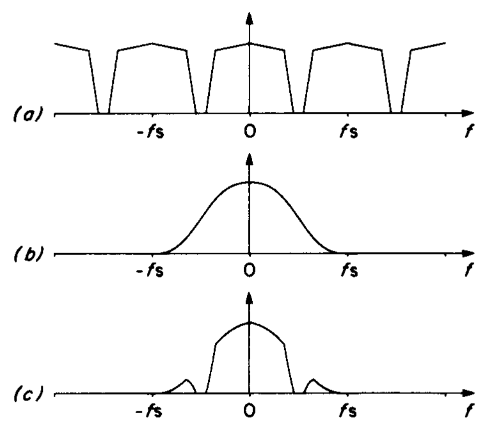


Fig. 7 – Spectra corresponding to the waveforms of Fig. 6: (a) the spectrum of a sampled signal, (b) the frequency characteristic of a filter with the impulse response shown in Fig. 6(b) and (c) the spectrum of the reconstituted waveform, obtained by multiplying (a) and (b).

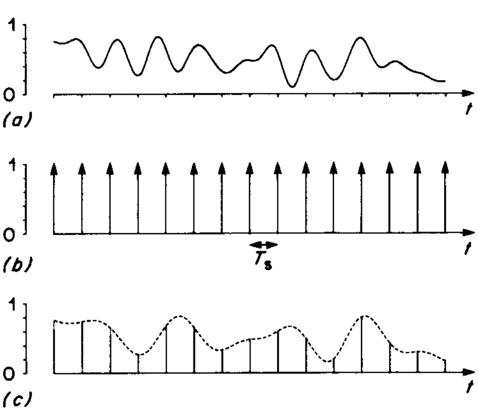


Fig. 8 – The effect of undersampling: (a) a signal waveform containing frequencies above half the sampling frequency, (b) the sampling impulses and (c) the weighted impulses representing the sample values. Returning these samples to continuous form using the ideal reconstituting filter (Fig. 5) produces the waveform shown dashed in Fig. 8(c).

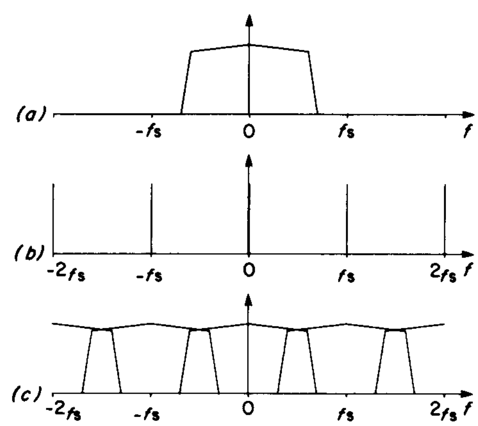


Fig. 9 – Spectra showing the effect of using too low a sampling rate: (a) the spectrum of a continuous signal with components beyond the half sampling frequency, (b) the spectrum of the sampling impulses, and (c) the spectrum of the sampled signal showing the overlap between adjacent spectra.

effect in the frequency domain, in which the baseband and repeated spectral components overlap. Such components, which have been shifted from their true positions in the normal band of signal frequencies, are known as alias components. In the overlap region, it is impossible to distinguish the true components of the baseband spectrum from the alias components of the repeated spectrum. Under these circumstances it is generally best to suppress all frequencies in the overlap band. This is because the loss of wanted frequencies is usually less noticeable than the presence of alias components.

2.2. Sample rate changing

2.2.1. Resampling

In a sampled waveform, such as Fig. 1 (c), the signal value is only known at the instants the samples were taken and is undefined in between. To change the sampling rate, it is necessary to obtain the signal value at new sampling points, between the original sample positions. One method would be to return the sampled waveform to continuous (analogue) form using a low-pass filter as described in the previous Section. Then the continuous signal could be resampled at the new rate in the normal way. If the reconstituted continuous signal is accurate, then resampling will not introduce any additional impairment. However, inaccuracies in the reconstituted waveform may be exaggerated by the resampling process.

For example, if the waveform of Fig. 3 (c) is resampled at a higher rate, interval T'_s, this gives new sample values as shown in Fig. 10. Frequency spectra corresponding to these waveforms are shown in Fig. 11. Here, the process of convolution causes unsuppressed remnants of the original sampling process to fall into the baseband region. The alias components, although noticeable in themselves, may interfere with the true signal to produce more noticeable beat patterns. Also, the low-pass filter used to return the original samples to a continuous waveform produces an attenuation of the baseband spectrum which affects the true components in the resampled spectrum.

If the new sampling rate is lower than the original rate, then it is possible for the spectrum of the reconstituted signal to extend beyond the new half sampling frequency, even though the original waveform has been reproduced accurately. Thus, the spectrum of the new samples would include an overlap region, similar to that of Fig. 9 (c). To avoid this, a further amount of low-pass filtering is required before resampling to limit the signal components to below the new half sampling frequency. This additional band-limitation can be incorporated as part of the action of the sample reconstitution low-pass filter.

2.2.2. The interpolation process

Although resampling of a reconstituted continuous waveform could be used for sample rate conversion, the intermediate continuous signal stage is unnecessary, as the value of each new sample can be calculated directly from the values of surrounding samples. This is known as interpo-

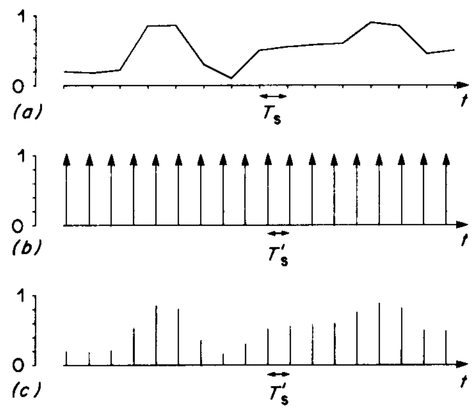


Fig. 10 – Resampling a previously sampled and reconstituted waveform: (a) the reconstituted waveform of Fig. 3 (c), (b) sampling impulses spaced at T'_s, a shorter interval than in the first sampling process, and (c) the resulting sample values.

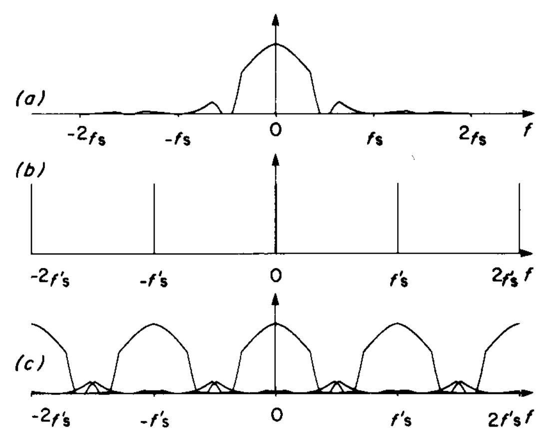


Fig. 11 – Spectra corresponding to the waveforms of Fig. 10: (a) the low-pass filtered spectrum of Fig. 4(c), (b) the spectrum of sampling impulses at a higher sampling frequency f'_s, and (c) the result of convolving (a) and (b). The residual harmonic spectra centred on f_s, $2f_s$, etc. in (a) are aliased to overlap the baseband spectrum in (c).

lation. In practice, the interpolation method is preferable because it is inherently digital, avoiding the intervening analogue process, but in principle the two methods are equivalent and are capable of identical results.

As a mathematical process, interpolation consists of assuming a particular form of continuous function, usually a polynomial expression, which passes through, or near to, one or more of the original sample values. This allows new sample values to be calculated for any points between the original sample positions. For example, with linear interpolation it is assumed that the function follows a straight line between sample points as shown in Fig. 12. The new value s depends, therefore, both on the two nearest sample values, s_1 and s_2, taken at t_1 and t_2 respectively, and on the position t of the new sample between the two; s can be calculated from the following expression:

$$s = c_1 s_1 + c_2 s_2$$

where

$$c_1 = 1 - \frac{t - t_1}{t_2 - t_1}$$

and

$$c_2 = \frac{t - t_1}{t_2 - t_1}$$

One form of interpolator, shown in Fig. 13, consists of registers R to hold the input sample values, $s_1, s_2, \ldots$, multipliers to form the products, $c_1 s_1$, $c_2 s_2, \ldots$, and adders to sum the products, $c_1 s_1 + c_2 s_2 + \ldots$ For linear interpolation only two

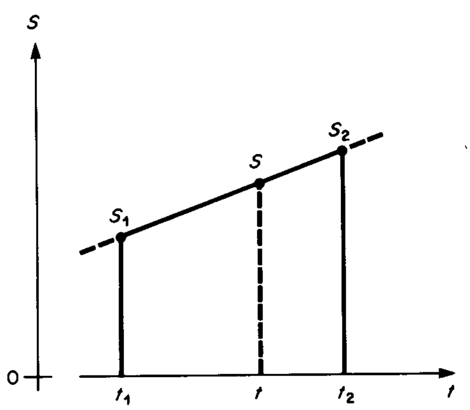


Fig. 12 – Linear interpolation: to obtain a sample value at time t, the shape of the waveform between the samples at t_1 and t_2 is assumed to follow a straight line.

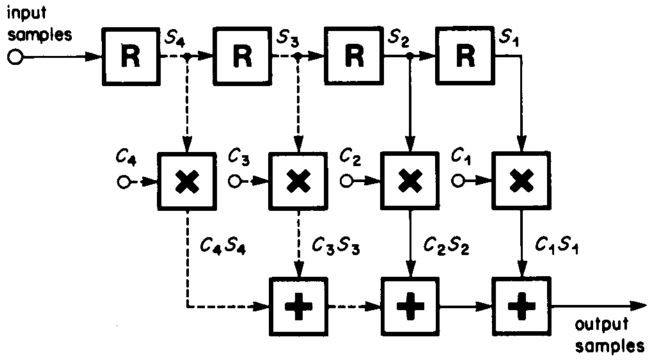


Fig. 13 – An extensible form of interpolator, consisting of registers (R), multipliers and adders.

stages are required and the coefficients are obtained from the expressions shown above for c_1 and c_2. Additional stages (shown dashed) can be added for more complicated interpolation methods.

The way in which the interpolation coefficients vary with the relative positions of the input and output samples can be expressed in the form of an interpolation aperture function. In general, this is a continuous function because all phases of input and output samples can occur when the input and output sampling rates are unrelated. The total extent of the function is known as the aperture or aperture width and is measured in input sampling periods.

Notionally, the aperture function is applied by placing its origin at a required new sample position so that the weighting coefficient value for each input sample can be selected according to its position in relation to the aperture function. This is shown in Fig. 14 for linear interpolation, which has a triangular interpolation aperture function as shown in Fig. 14(b). In this case, the samples produced by interpolation in Fig. 14(c) are identical to the samples formed by low-pass filtering and resampling in Figs. 3 and 10. This is because the interpolation aperture function, Fig. 14(b), and the low-pass filter impulse response, Fig. 3(b), have the same shape.

In the frequency domain, the interpolator performance is determined by a frequency characteristic, which can be found by evaluating the Fourier transform $G(f)$ of the interpolation aperture function $g(t)$:

$$G(f) = \int_{-\infty}^{\infty} g(t)\,\mathrm{e}^{-2\pi \mathrm{j} f t}\,dt$$

Conversely, the aperture function corresponding to any desired spectral characteristic can be obtained

by the inverse Fourier transform:

$$g(t) = \int_{-\infty}^{\infty} G(f)\, e^{2\pi j f t}\, df$$

For linear interpolation, the aperture function can be defined, by inspection of Fig. 14(b), as:

$$\begin{aligned} g(t) &= t + T_s & -T_s \leqslant t \leqslant 0 \\ &= T_s - t & 0 < t \leqslant T_s \\ &= 0 & |t| > T_s \end{aligned}$$

Then

$$G(f) = \int_{-T_s}^{0} (t + T_s)\, e^{-2\pi j f t}\, dt + \int_{0}^{T_s} (T_s - t)\, e^{-2\pi j f t}\, dt$$

which after integration simplifies to:

$$G(f) = \frac{\sin^2 (\pi f T_s)}{(\pi f)^2}$$

This spectral characteristic for linear interpolation is shown in Fig. 15 along with the spectra for the input and output samples; these correspond to the time domain waveforms in Fig. 14.

As the filter characteristic of linear interpolation is the same as that shown in Fig. 4(b), the action of the interpolator is equivalent to that shown in Figs. 4 and 11, although the signal never exists in an intermediate continuous form. Indeed, it is useful to visualise the interpolation process as consisting of an equivalent low-pass filtering stage, followed by resampling. Then the performance of the interpolation method is still governed by the efficacy of its filtering action in suppressing unwanted harmonic components and retaining the baseband spectrum.

2.2.3. Synthesis of aperture functions

Although the Fourier transform provides a method of calculating the performance of any aperture function, it is preferable to design an interpolator by specifying its performance directly in the frequency domain. However, in most cases, this would produce an aperture function of unlimited width, so requiring the interpolation process to include an infinite number of samples. A truncated version could be used, but then the frequency characteristic would differ from that originally specified. What is required is a method of specifying the frequency characteristic in a manner that is constrained to produce an aperture of known width.

Such a method[5,10] consists of specifying the frequency characteristic at intervals of $(1/k)f_s$, in the knowledge that this will produce an aperture function of width k sample periods. This can be demonstrated by considering an aperture function of width kT_s as the product of a block pulse of width kT_s and a repetitive waveform of period kT_s. Fig. 16 shows an example of such a 'fixed-point' aperture with a width of four sample periods.

The frequency spectra corresponding to these waveforms are shown in Fig. 17: the spectrum of

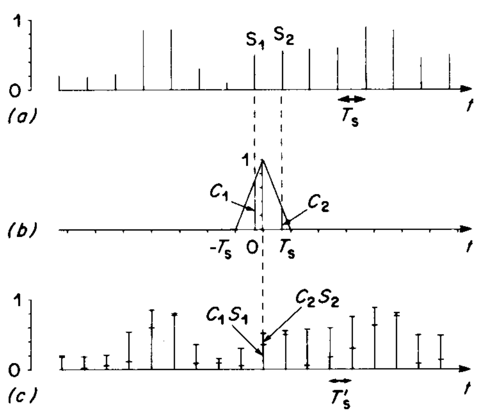


Fig. 14 – The interpolation process: (a) the input samples, (b) the interpolation aperture function, showing the coefficient values selected to multiply the input samples and (c) the new sample values, each produced by combining weighted contributions from two input samples.

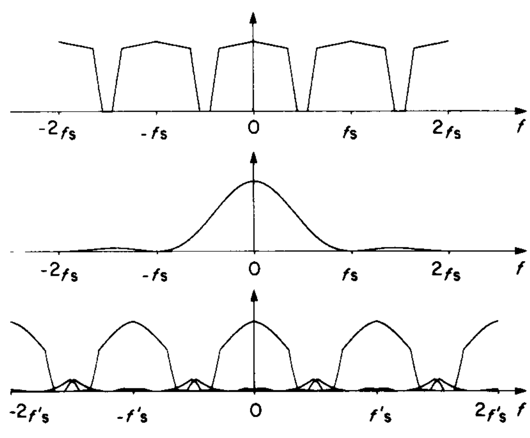


Fig. 15 – Frequency spectra for the interpolation process: (a) the spectrum of the input samples, (b) the frequency characteristic for linear interpolation and (c) the spectrum of the new samples.

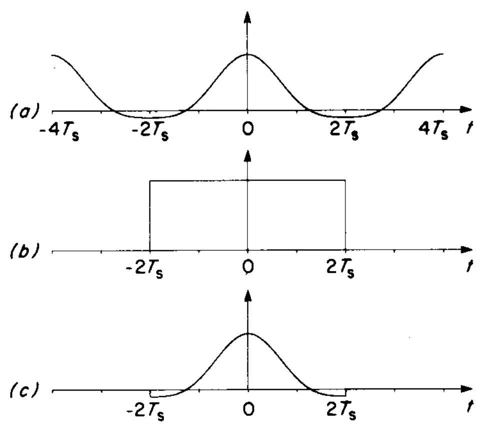


Fig. 16 – Aperture functions of set width: An aperture function of width $4T_s$, shown at (c), can be considered as the product of (a) a repetitive waveform of period $4T_s$ and (b) a block pulse of width $4T_s$.

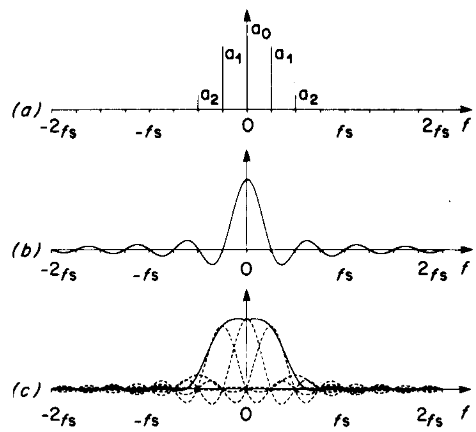


Fig. 17 – Spectra corresponding to the waveforms of Fig. 16: (a) a series of line spectra spaced at intervals of $\frac{1}{4}f_s$, (b) a sinc function with period $\frac{1}{4}f_s$ and (c) the frequency characteristic (solid line) produced by convolving (a) and (b); the individual contributions are shown dashed.

the repetitive waveform is a series of line spectra, Fig. 17(a), spaced at intervals of $(1/k)f_s(k = 4)$, with amplitudes a_0, a_1, a_2, ... representing the Fourier series components of the waveform; the spectrum of the block pulse is a sinc function, Fig. 17(b), with its first zero at $(1/k)f_s$; then the frequency characteristic corresponding to the aperture function is produced by convolution of the line spectra with the sinc function as shown in Fig. 17(c). Therefore, specifying the values $a_0, a_1, a_2, \ldots$ in the frequency domain produces an aperture function of width k sample intervals, which can be calculated from the Inverse Fourier transform of the pairs of line spectra shown in Fig. 17(a):

$$\frac{2}{k}\left(a_0 + 2a_1 \cos\frac{2\pi t}{kT_s} + 2a_2 \cos\frac{4\pi t}{kT_s} + \cdots\right)$$

Any number of values can be specified, but usually all values beyond $\pm\frac{1}{2}f_s$ will be set to zero as indicated in Fig. 17(a). It should be noted that although asymmetrical frequency characteristics could be generated by having asymmetrical aperture functions, this would result in unwanted phase shifts in the interpolated signals.

The aperture function, Fig. 16(c), produced by specifying fixed-point values in the frequency domain has the same width, four sample periods, as the smoothed sinc function impulse used for low-pass filtering in Fig. 6. However, in Fig. 18, when the fixed-point aperture (Fig. 18(b)) is used to reconstitute the samples from Fig. 1(c), the resulting continuous waveform (the solid line in Fig. 18(c)) generally provides a closer approximation to the original waveform (the dashed line in Fig. 18(c)) than that obtained in Fig. 6. When the frequency characteristics of the two filters are compared, Figs. 7(b) and 19(b), it is apparent that the fixed point method has produced a characteristic closer to the ideal rectangular shape, Fig. 5(a). Because of this, the filter provides greater rejection of harmonic spectra in the region $\frac{1}{2}f_s$ to f_s, while introducing little in-band attenuation. The fixed-point method therefore provides a simple and effective means of synthesising interpolation aperture functions.

2.2.4. Aperture quantisation

An interpolation aperture function is continuous and therefore has an infinite number of values. In order to store the function in a read-only memory for digital interpolation, it is necessary to produce an approximation to the function, quantised in both the abscissa and ordinate directions. Thus, the function value is only specified at a limited number of positions and also each stored value will have only limited accuracy.

Read-only memories commonly have 2^n locations so it is convenient to store the value of the aperture function at $N = 2^n$ equi-spaced points. This even number has the additional advantage that the two end points of the function cannot both contribute at once – a situation which would require an extra multiplier.

Fig. 20 shows the effect of representing the linear interpolation aperture function at only eight points (amounting to four values per input sample

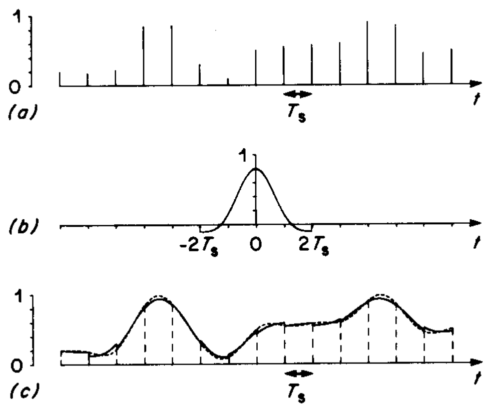


Fig. 18 – Signal reconstitution using the 'fixed-point' filter of Fig. 17: (a) the sampled signal of Fig. 1(c), (b) the impulse response of the fixed-point filter and (c) the reconstituted waveform (solid line); the original waveform and the sample impulses are shown dashed for comparison.

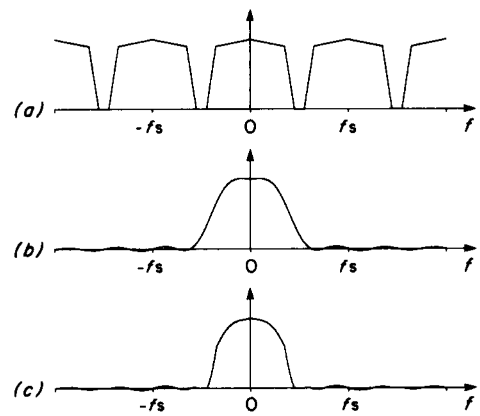


Fig. 19 – Signal reconstitution in the frequency domain using the filter of Fig. 17: (a) the spectrum of a sampled waveform, (b) the frequency characteristic of the fixed-point filter and (c) the spectrum of the reconstituted signal.

period). The eight values shown in Fig. 20(a) are stored, but each value is used over a region of $\pm\frac{1}{8}T_s$, as shown in Fig. 20(b). The effective aperture function is therefore the piecewise constant approximation shown in Fig. 20(c) which results from the convolution of Figs. 20(a) and (b).

In the frequency domain, sampling of the aperture function at an interval of $\frac{1}{4}T_s$ causes the frequency characteristic of the continuous aperture function, Fig. 15(b), to be repeated at harmonics of $4f_s$, as shown in Fig. 21(a). This spectrum is then multiplied by the spectrum of the block pulse Fig. 21(b) (equivalent to convolution in the time domain) which attenuates the repeated spectra as shown in Fig. 21(c). When more aperture values are stored, the repeated spectra in Fig. 21(a) are displaced further from the baseband spectrum and are more attenuated by the action of the block pulse spectrum, Fig. 21(b). Table 1 shows values of the maximum attenuation of wanted frequencies (over the band 0 to $\frac{1}{2}f_s$) and the minimum attenuation of unwanted frequencies (in the band $(N-\frac{1}{2})f_s$ to Nf_s) for $N=2$, 4, 8, 16 and 32 values per sample period. The higher values of N attenuate the unwanted spectra sufficiently to reduce aliasing to a level considered acceptable in the unsampled characteristic.

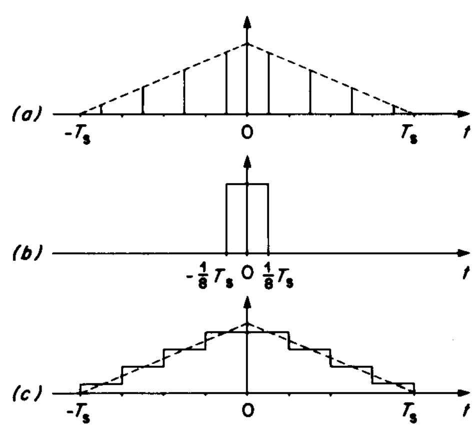


Fig. 20 – Time quantisation of aperture functions: (a) eight impulse values spaced at intervals of $\frac{1}{4}T_s$ which could be stored and used to represent the linear interpolation aperture function (shown dashed), (b) an interval of $\pm\frac{1}{8}T_s$ over which each value is used and (c) the time-quantised aperture function produced by convolving (a) and (b).

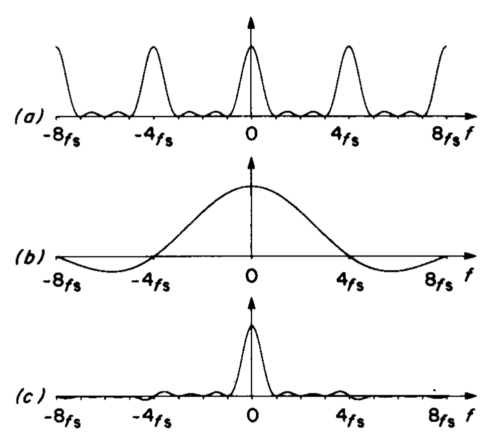


Fig. 21 – Spectra corresponding to the time functions of Fig. 20: (a) the frequency characteristic of the sampled linear interpolation aperture, (b) the sinc function spectrum of the block pulse of Fig. 20(b), and (c) the frequency characteristic of the time-quantised aperture function, produced by multiplying (a) and (b).

Table 1

N	Wanted band maximum attenuation	Unwanted band minimum attenuation
2	0.637 (-4dB)	0.300 (-10dB)
4	0.974 (-0.2dB)	0.139 (-17dB)
8	0.994 (-0.05dB)	0.066 (-24dB)
16	0.998 (-0.02dB)	0.032 (-30dB)
32	1.000 (-0.01dB)	0.016 (-36dB)

The attenuated sampling products centred on Nf_s are aliased back into the baseband region by the resampling effect of interpolation in the same way as shown for the signal sampling products in Fig. 15(c).

In addition to the aperture function being defined only at certain positions, the function values at those positions are also quantised. This is because the coefficients are stored and used as binary numbers so that their accuracy is limited by the number of bits available. Whereas it is straightforward to store the coefficients to high accuracy, the limiting factor is usually the capacity of the multipliers in the interpolator.

Quantisation of the values alters the effective aperture function and so changes the frequency characteristic. Direct quantisation of the individual function values, rounding each to the nearest allowed value, may cause the sets of coefficients used for each input–output sampling offset no longer to sum to unity. This would cause a variation in gain as the coefficients were changed from one set to another. In the frequency domain, the gain variation results because zeros in the frequency characteristic no longer fall at the centres of the repeated spectra.

The problem can be avoided by ensuring that the sets of coefficients used for each offset position of the aperture still sum to unity after quantisation. An effective algorithm for quantisation consists of first rounding down all the coefficients of a set and summing the results. If the sum is less than unity, sufficient coefficients are rounded up to make up the difference. The overall error is minimised by selecting coefficients in which rounding up will produce the minimum error relative to the actual value of each coefficient; even so, this may cause quantisation errors greater than half a least significant bit in individual coefficients.

The effects of quantisation can be demonstrated by taking the time quantised version of the linear function, Fig. 20(c), and quantising this in value very coarsely, so that only the values $\frac{3}{4}$ and $\frac{1}{4}$ are allowed as shown in Fig. 22(a). For any position of an output sample, one input sample is multiplied by $\frac{3}{4}$ and the other by $\frac{1}{4}$, so the requirement that each set of coefficients should sum to unity is satisfied. This modifies the frequency characteristic to the form shown in Fig. 22(b) as a solid line. The original characteristic for linear interpolation is shown dashed for comparison. Even with this very coarse degree of quantisation, the frequency characteristic is not greatly altered. In practice, it is convenient to use 7- or 8-bit accuracy in the coefficients, so that the characteristic for the quantised aperture function is usually very close to that of the continuous function.

Quantised aperture functions, such as those shown in Figs. 20(c) and 22(a), can be considered to be made up of a series of elements comprising symmetrical pairs of block pulses added together. Each pair has the basic form shown in Fig. 23, with amplitude A and width T centred on $\pm t_0$. Fourier transformation of this aperture element reveals that the associated component of the frequency

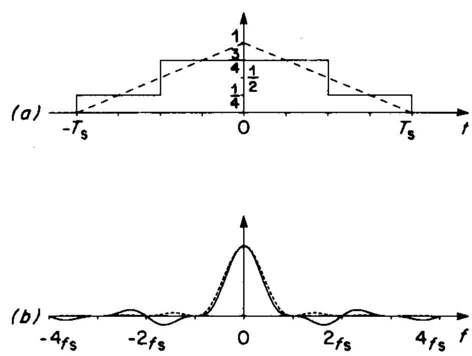


Fig. 22 – Amplitude quantisation of aperture functions: (a) the aperture function of linear interpolation (shown dashed) coarsely quantised to the values $\frac{1}{4}$ and $\frac{3}{4}$ (solid line) and (b) the frequency characteristic of the quantised aperture function (solid line) compared with that of the original aperture function (shown dashed).

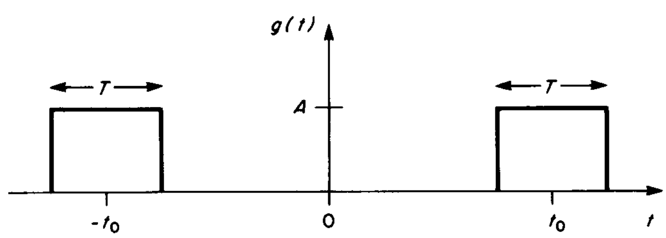


Fig. 23 – A symmetrical pair of block pulses representing a generalised element of a quantised aperture function.

characteristic is given by:

$$G(f) = 2AT\cos 2\pi t_0 f \cdot \frac{\sin \pi Tf}{\pi Tf}$$

Therefore, the frequency characteristic of a quantised aperture function can be calculated as a summation of such terms. When the block pulses are all of width T, the sinc term is common to all the elemental components.

3. Vertical–temporal interpolation of television signals

3.1. Television scanning

The normal application of the one-dimensional sampling process described in Section 2.1 is to convert a continuous waveform to a series of samples for p.c.m. coding. Although this form of sampling is often applied to television signals, the process of scanning used to originate television signals from an image is also a sampling process. Conversion from one scanning standard to another is, therefore, a sampling rate conversion and follows the same basic theory described in Section 2.2.

The brightness of an image falling onto a television camera tube is a function varying in three dimensions: horizontal position (x), vertical position (y) and time (t). The action of scanning the image with the television raster pattern converts this three-dimensional function into a one-dimensional signal.

The simplest form of scanning used to produce a television signal is sequential scanning. A 312 lines per picture, 50 fields per second sequential scan is shown diagrammatically in Fig. 24(a). Each consecutive set of lines is taken at the same vertical positions, but is offset in time from the previous set by one field period. Although not shown in the diagram, the columns of scanning points are very slightly skewed from the vertical, so that the last line of one field scan is taken just before the first line of the next. Similarly the lines themselves are not precisely horizontal; (in the diagram, the horizontal dimension extends into the paper).

All broadcast television systems, however, use interlaced scanning. This is shown in Fig. 24(b) for a 625 lines per picture, 50 fields per second, interlaced scan. In this case, the lines of successive fields are interleaved so that each line falls in a position vertically between those on the adjacent fields. This doubles the number of lines in the picture, while keeping the scanning rates virtually the same.

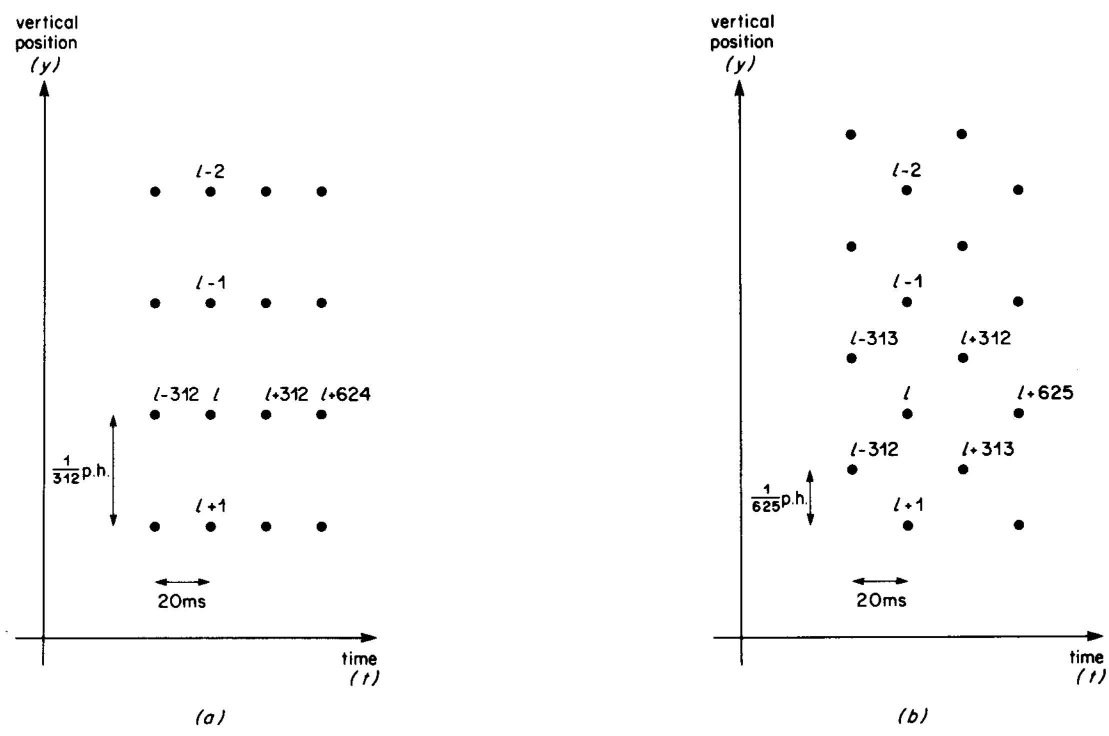


Fig. 24 – Television scanning: diagrammatical representations of (a) the pattern of lines for a 312 lines per picture, 50 fields per second sequential scan and (b) a 625 lines per picture, 50 fields per second interlaced scan. The horizontal dimension extends into the paper.

In the scanned signals the horizontal dimension is still continuous, but the image brightness is now only defined at discrete intervals in the vertical and temporal dimensions. Thus, the image brightness has been sampled in the two dimensions of vertical position and time.

The action of sampling repeats the frequency spectrum of the image at harmonics of the sampling frequency, as described in Section 2.1. The image spectrum consists of horizontal frequency (m), vertical frequency (n) and temporal frequency (f) components. Horizontal frequencies in the image are not affected by scanning because the horizontal dimension of the signal remains continuous. However, in a digital standards converter, the horizontal components are also sampled; but, provided that a line-locked sampling frequency is used, this sampling is orthogonal to the vertical–temporal sampling resulting from scanning. Because of this, the two processes have no effect on one another and can be treated completely separately. Therefore, only the vertical and temporal components of the image spectrum, represented diagrammatically in Fig. 25, are affected by scanning. Temporal frequencies are measured in hertz (Hz), whilst the units of vertical frequency are cycles per picture height (c/p.h.).

If the scene from which the image is formed is completely still, then the frequencies which make up the image spectrum have no temporal component and all the spectral energy lies along the vertical frequency axis. If an object in the scene appears and disappears at a regular rate, such as a light flashing, this results in temporal frequency components which are positioned away from the n axis by an amount depending on the rate of flashing. However, most temporal frequencies are

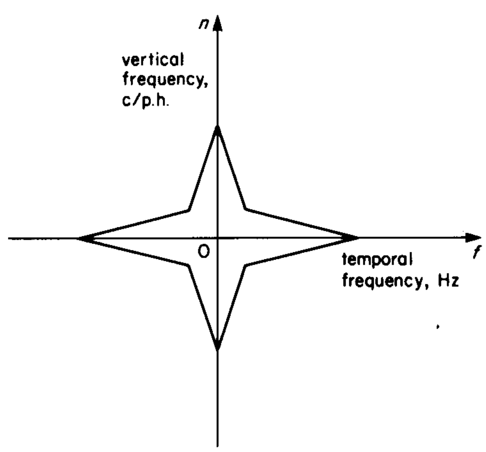


Fig. 25 – A diagrammatical representation of the vertical–temporal frequency spectrum of an image.

produced by the movement of objects in the scene. The effect of movement is more complicated because the temporal frequencies produced depend on both the rate of movement and the spatial frequencies which make up the object.

For simplicity, consider first an object consisting of a single spatial frequency component, a sine wave, Fig. 26(a). As this moves across a point in the scene, it produces a single temporal frequency, Fig. 26(b), corresponding to the time it takes to move through its own spatial period. If the rate of movement is increased or if an object of greater spatial frequency is used with the same rate of movement, the temporal frequency resulting as the object passes a point will be increased.

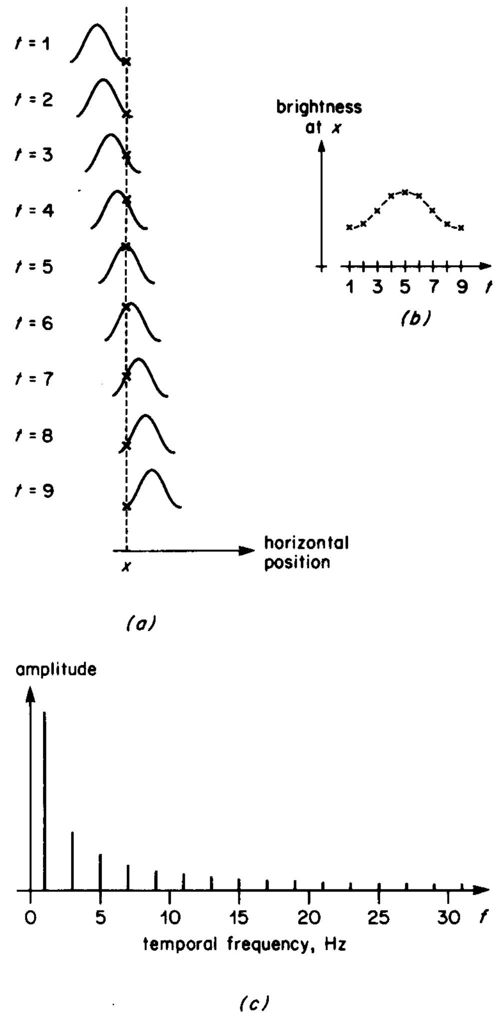


Fig. 26 – Temporal frequency produced by a moving object: (a) an object with a sinusoidal brightness profile passing a position x, shown at several equi-spaced time intervals, and (b) the waveform of brightness at position x, consisting of a single temporal frequency. (c) Temporal frequencies produced by a square wave object moving through its own spatial period in one second.

More complicated shapes can be considered as a series of Fourier components. Thus, if the highest spatial frequency in the object is S times the fundamental component, as the object moves it will produce a series of temporal components. These will extend to S times the basic temporal frequency corresponding to the object moving through its own width. Fig. 26 (c) shows the temporal frequencies resulting from a square wave moving through its own spatial period in one second. In this case, the presence of high spatial frequencies in the square wave results in high temporal frequencies, even with a very modest rate of movement. Thus, sharp pictures produce a greater proportion of high temporal frequencies than soft pictures. Conversely, reducing the spatial frequency components in the signal, for example by including a horizontal low-pass filter, makes a corresponding reduction in those temporal frequencies produced by horizontal movement. Temporal frequencies produced by a stationary object flashing would not be affected.

For purely horizontal movement, all the spectral energy is concentrated along the f axis of the n–f spectrum. If, instead, an object is moving vertically, this produces a series of temporal components along a diagonal line passing through the origin; the angle made between this line and the n axis increases with the speed of movement of the object.

Clearly there is virtually no limit to the spatial and temporal frequencies that can occur in a scene, but some filtering is introduced by the action of the camera. The camera tube integrates the light falling on it at each point for a field period, thus attenuating the higher temporal frequencies. (It is generally assumed that interlaced line positions are discharged on each field scan.) Also, the width of the scanning spot is not infinitely small, so that the charge from adjacent areas of the tube surface contributes to the output current, thereby filtering the image vertically. This gives a probable form of camera filtering response, including aperture correction, as shown in Fig. 27.*

The effect of scanning is to repeat the spectrum of the image of the original scene, Fig. 25, filtered by the characteristic of Fig. 27, at harmonics of the scanning rates. The 312 lines per picture, 50 fields per second sequential scan produces the spectrum shown in Fig. 28 (a), while interlaced scanning, with 625 lines per picture and 50 fields per second, produces the spectrum shown

*This model for camera response was derived by J. O. Drewery.

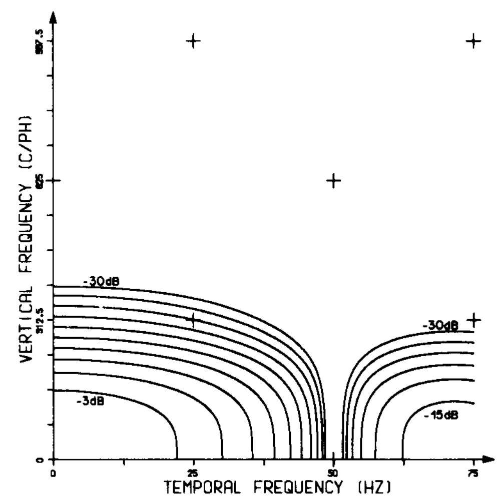


Fig. 27 – The probable form of the vertical and temporal filtering effect of a television camera, including vertical aperture correction. The characteristic shows contours of constant attenuation at 3 dB intervals down to −30 dB.

in Fig. 28 (b). Although only the first quadrant is shown, the spectra extend into the other quadrants in the same pattern.

As components of the spectra can extend well beyond the outlines shown in Fig. 28, aliasing forms an accepted part of normal television pictures. For example, with interlaced scanning, flicker arises from vertical frequency components in the region around $(0, 312\frac{1}{2})$ in the image spectrum. In the scanned spectrum these components are repeated around $(25, 0)$, arising from the spectra centred on $(25, \pm 312\frac{1}{2})$. So the vertical detail flashes with a temporal frequency of 25 Hz.

When Figs. 27 and 28 are considered together, it is evident that, for the interlaced system, temporal aliasing from camera pictures is much more likely than vertical aliasing. However, electronically generated signals are, in general, completely unfiltered, temporally and vertically. Because of this, these signals often have a much greater degree of aliasing than can occur with signals from a camera. Therefore, captions and graphics are usually considerably more difficult to convert successfully from one standard to another than normal camera pictures. Similarly, signals derived from scanning film in a telecine tend to produce more vertical resolution, and aliasing, than a television camera. Also, film pictures already include severe temporal aliasing resulting from the 25 (or 24) frames per second temporal sampling action of the film camera.

Spectra for other scanning standards would, in general, be similar to those of Fig. 28 with

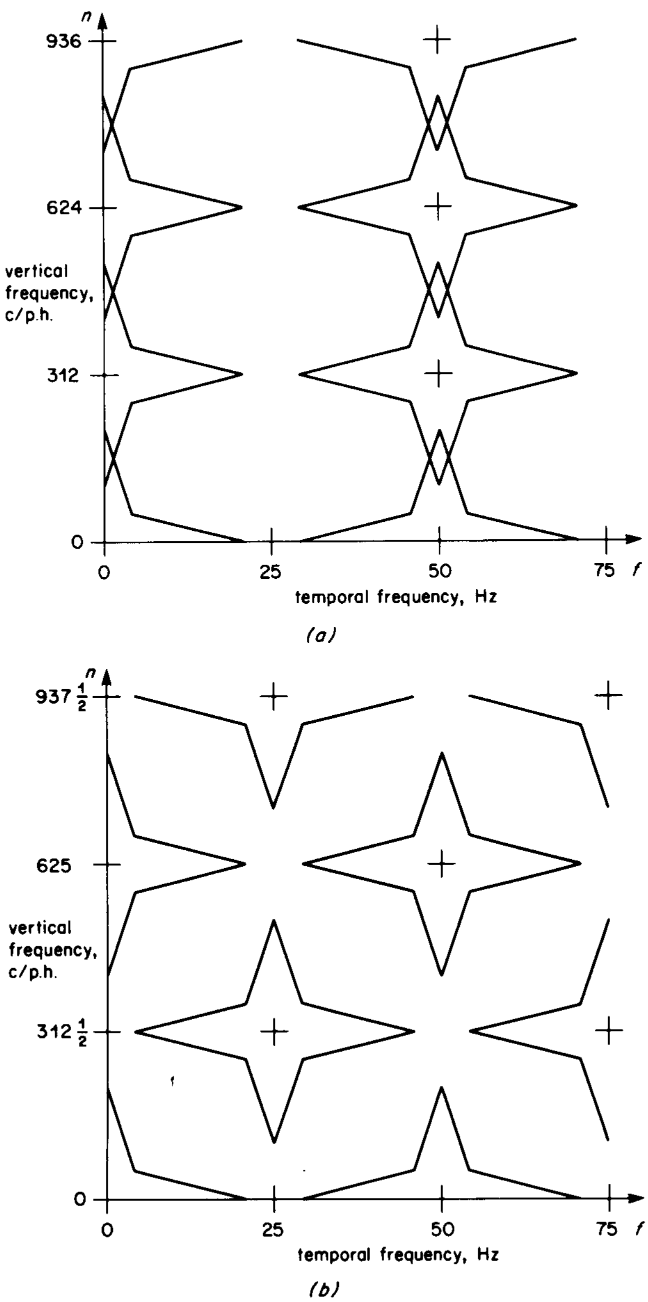


Fig. 28 – Television scanning in the frequency domain: the spectrum of the image (Fig. 25) is repeated at harmonics of the scanning rates (a) by a 312 lines per picture, 50 fields per second sequential scan and (b) by a 625 lines per picture, 50 fields per second interlaced scan.

appropriate scaling of the axes. In particular, for the 525 lines per picture, 60 fields per second, interlaced standard, the repeated spectra would be centred on $(0, 525)$, $(30, 262\frac{1}{2})$, $(60, 0)$, $(60, 525)$, etc. Also, the filtering effect of cameras would be a scaled version of Fig. 27.

Ultimately the television signal is reconstituted in its three-dimensional (x, y and t) continuous form for viewing. This is achieved by scanning the brightness information onto the television cathode ray tube display. Although the display tube provides some low-pass filtering of vertical and temporal frequencies (through the profile of the scanning spot and phosphor, and the time constant of the phosphor), it has a relatively slow roll-off so that the harmonic spectra are poorly suppressed. The presence of these extra components results in the visibility of the line structure for vertical components and the presence of large-area flicker for temporal components.

With interlaced scanning, the vertical repeated spectra are separated from the baseband spectrum by a wider margin. Because of this, it is possible to display a reasonable range of vertical frequencies and still to suppress the line structure. The relatively low vertical resolution provided by cameras (Fig. 27) suggests that the extra resolution possible with interlaced scanning is largely unused. If it were fully used, this would result in a substantial increase in interlace flicker. It should be concluded, therefore, that interlacing is primarily a means of suppressing the line structure, rather than a method of increasing vertical resolution.

3.2. Conversion methods

Before electronic standards converters were available, optical converters were used consisting of a display tube and a television camera. The conversion was made by displaying the input standard picture in the normal way and scanning the reconstituted image with a camera using an output standard raster. In terms of sample rate conversion, this process is equivalent to the low-pass filtering and resampling method described in Section 2.2.1. In addition to problems of geometric distortion caused by imperfect scanning, the low-pass filtering action of an optical converter is difficult to control, since it is the result of phosphor characteristics and spot profiles. Because of this, present day electronic converters achieve better performance. Nevertheless, it is still useful, as described in Section 2.2.2, to visualise the interpolation process of an electronic converter as consisting of low-pass filtering to recover the baseband spectrum of the original image, followed by resampling on the new standard.

For interlaced scanning, the vertical–temporal low-pass filter characteristic needs to have a shape approximating that shown in Fig. 29. This triangular shape would retain the main components of the baseband spectrum and reject the repeated spectra centred on multiples of the scanning rates.

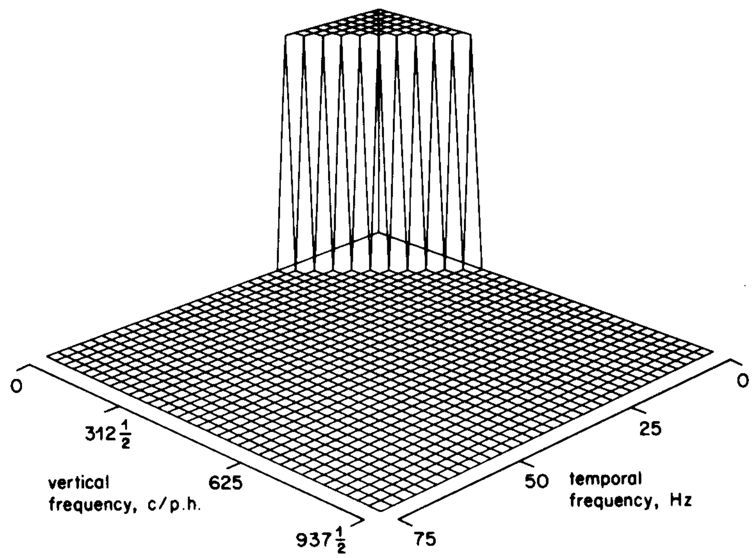


Fig. 29 – The triangular shape of the idealised vertical–temporal low-pass filter characteristic needed to retain the baseband spectrum and reject the harmonic spectra resulting from interlaced scanning.

Because television scanning is a two-dimensional sampling process, it is possible to treat vertical and temporal (line and field) interpolation either as two one-dimensional processes or as a single two-dimensional process. This Section examines methods of using two separate interpolation processes and demonstrates in the frequency domain the drawbacks of this approach when applied to interlaced scanning. This is then compared with the greater flexibility and improved performance of using a single, two-dimensional interpolation process.

3.2.1. Separate vertical and temporal interpolation

In a converter that uses separate vertical and temporal interpolation, the two-dimensional interpolation aperture function $g(t, y)$ is the convolution of two one-dimensional aperture functions, $g_1(t)$ and $g_2(y)$, one exclusively a function of time and the other exclusively a function of vertical position:

$$g(t, y) = g_1(t) * g_2(y)$$

Similarly, the overall two-dimensional frequency characteristic is the product of two one-dimensional characteristics:

$$G(f, n) = G_1(f) \cdot G_2(n)$$

Such two-dimensional functions are said to be variables-separable. Whenever either of the one-dimensional frequency characteristics (G_1 or G_2) has a zero value, the two-dimensional product characteristic (G) has a line of zero values, parallel to the other axis. This causes all variables-separable interpolation methods to have approximately rectangular frequency characteristics.

For sequential scanning, the rectangularly shaped characteristics of separate vertical and temporal interpolation are ideal. In this case, the interpolation processes can, in principle, be carried out in either order, as shown in Fig. 30, with equivalent performance. However, in practice, in a digital converter it is advantageous to convert the field rate first (Fig. 30(a)). This is because the results of the first interpolation need to be represented to higher accuracy to minimise rounding errors; storing these higher accuracy digital words in the smaller capacity line stores minimises the overall storage requirements.

The two separate interpolation processes are illustrated in Fig. 31. In the first step, Fig. 31 (a), lines of the intermediate standard are interpolated at a new field position; each of these new lines is produced from several lines at the same vertical position taken from neighbouring fields. The second step, Fig. 31 (b), consists of interpolating between the lines of the intermediate standard to produce lines at the correct vertical positions for the output standard.

In the frequency domain, the filtering effect of the temporal interpolator needs to reject all the temporal repeated spectra, leaving the vertical harmonic components unaffected. Ideally, therefore, all components in the shaded region of Fig. 32 (a) would be suppressed. The resampling action of interpolation then introduces new temporal harmonic components at the positions shown in Fig. 32 (b). Similarly, the vertical interpolation, which follows, first suppresses all the vertical repeated spectra, Fig. 32 (c), and then introduces new harmonic spectra, as shown in Fig. 32 (d). In this 'down conversion' process, the filter needs to cut at the output standard vertical half sampling

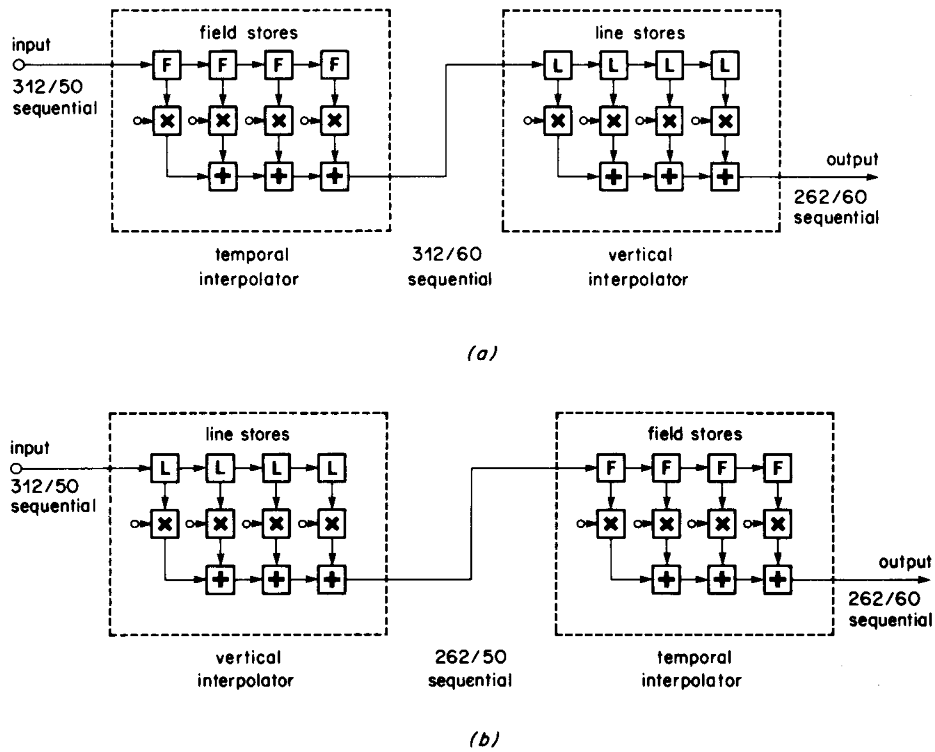


Fig. 30 – Converter arrangements for sequentially scanned signals in which vertical interpolation and temporal interpolation are carried out as separate processes: (a) temporal followed by vertical interpolation and (b) vertical followed by temporal interpolation.

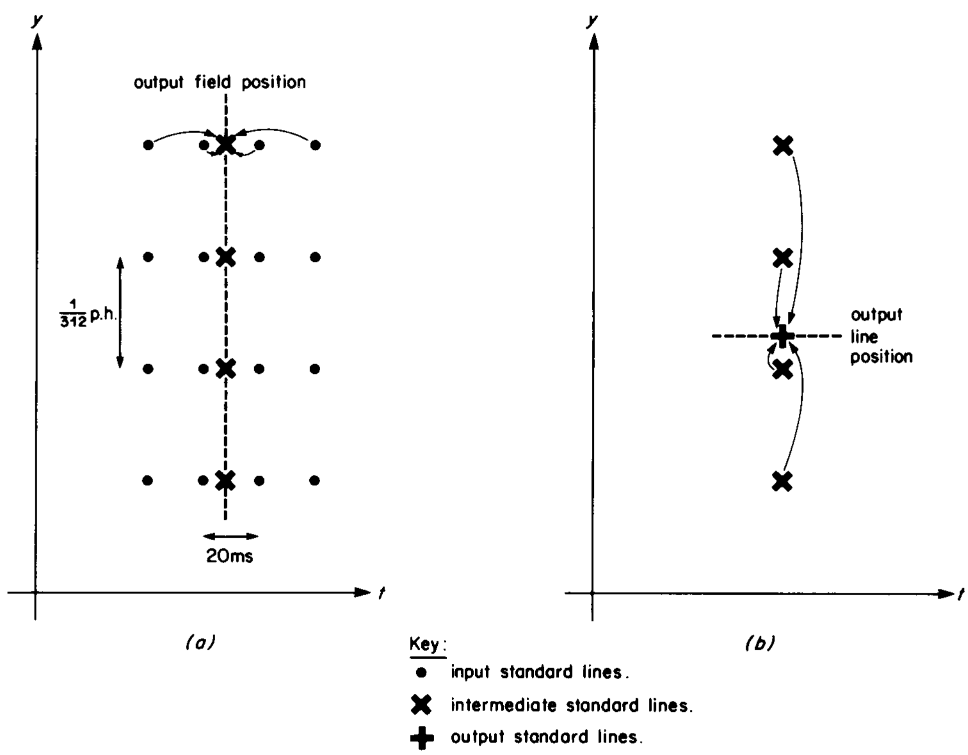


Fig. 31 – Temporal followed by vertical interpolation for sequential scanning: (a) lines of the intermediate standard are interpolated for a new temporal position, but at the same vertical positions as the input lines and (b) these intermediate standard lines are used to produce the output standard lines at new vertical positions.

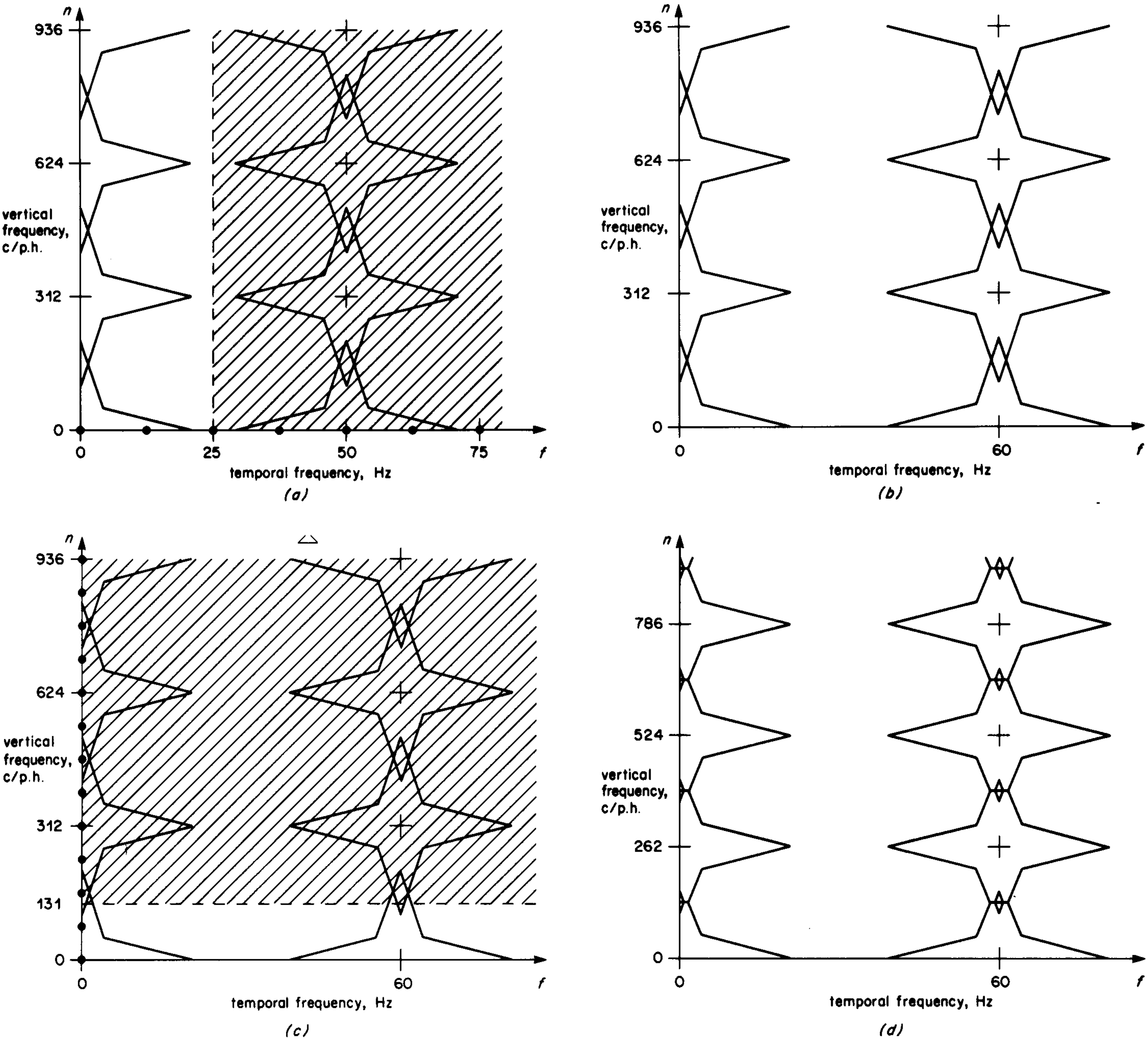


Fig. 32 – A spectral representation of temporal interpolation followed by vertical interpolation for sequential scanning: (a) first, temporal repeated spectra are rejected by the low-pass filtering action of the temporal interpolator (with a four-field aperture, the filter characteristic can be set independently at 12½ Hz intervals as marked on the temporal frequency axis), (b) the resampling action of the interpolator substitutes repeated spectra at the new field scanning rate, (c) then the vertical interpolator rejects the vertical repeated spectra (fixed points at 78 c/p.h. for a four-line aperture) and (d) substitutes repeated spectra with the new line spacing.

frequency, 131 c/p.h., to prevent the spectra overlapping on the output standard.

With interlaced scanning, applying separate vertical and temporal interpolation to the offset structure presents many problems. With separate interpolation, it is incorrect to interpolate using a mixture of lines from odd and even fields, either temporally or vertically. This is because a one-dimensional interpolation process cannot accommodate the two-dimensional offset of interlaced lines. For example, although vertical interpolation using lines from both types of fields would give good performance for still pictures, any movement

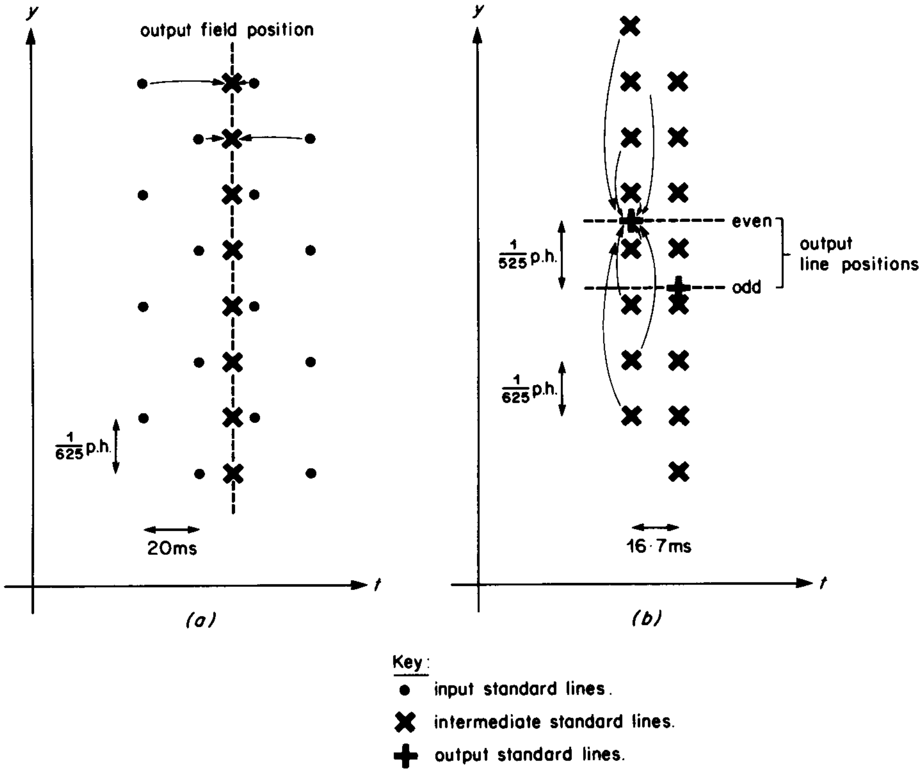


Fig. 33 – Temporal interpolation followed by vertical interpolation for interlaced scanning: (a) lines of a sequential intermediate standard are interpolated for a new temporal position, but at vertical positions corresponding to those of both even and odd input fields; (b) the intermediate standard lines are used to produce lines at new vertical positions appropriate for the even and odd positions on alternate output fields.

would result in serious impairments because the temporal offset between the fields would be ignored. With this restriction, separate interpolation can be applied to interlaced signals in either order. Whichever order is used, the first interpolation process must produce a higher sequential intermediate standard, in order to avoid interlace components which would interfere with the second interpolation process.

The action of an interpolator in which temporal interpolation precedes vertical interpolation for interlaced scanning is illustrated in Fig. 33. The temporal interpolator produces new fields at each output standard field position, as shown in Fig. 33(a), using input lines at the same vertical positions. Two sets of lines are interpolated for each new field position, corresponding to the positions of the odd and even field-lines of the input standard. Therefore, in the example used in the diagram, the 625/50 interlaced input standard is converted to a 625/60 sequential intermediate standard.

The second interpolator uses lines (eight lines in this example) from one field of the intermediate standard to produce new lines at positions appropriate for the output standard, as shown in Fig. 33(b). This includes the vertical offset from one field to the next required for interlacing. The vertical interpolator, therefore, converts the 625/60 sequential intermediate standard into the required 525/60 interlaced output standard.

Fig. 34 shows a block diagram of an interpolator arrangement which could perform the operations shown in Fig. 33. Note that two lines are produced by the temporal interpolator for each line of the input standard, resulting in an intermediate standard with twice the resolution of the input. For the reverse direction of conversion, that is, 525/60 to 625/50, the intermediate standard produced by this method would be a 525/50 sequential scan.

Fig. 35 illustrates the process of Fig. 33 in the frequency domain. In Fig. 35(a), the temporal interpolator attempts to suppress all the temporal repeated spectra; but suppression of the interlaced components centred on $(25, 312\frac{1}{2})$ proves impossible without rejecting most of the wanted baseband components as well. The compromise of using a cut-off at 25 Hz retains the wanted

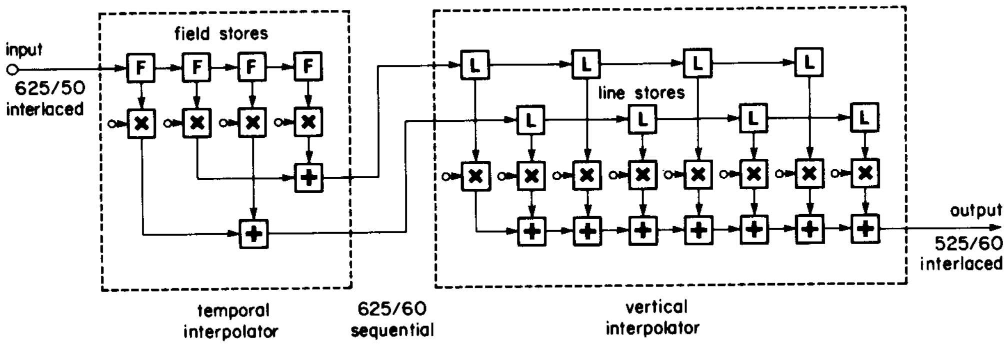


Fig. 34 – A converter arrangement for interlaced scanning in which temporal interpolation is followed by vertical interpolation.

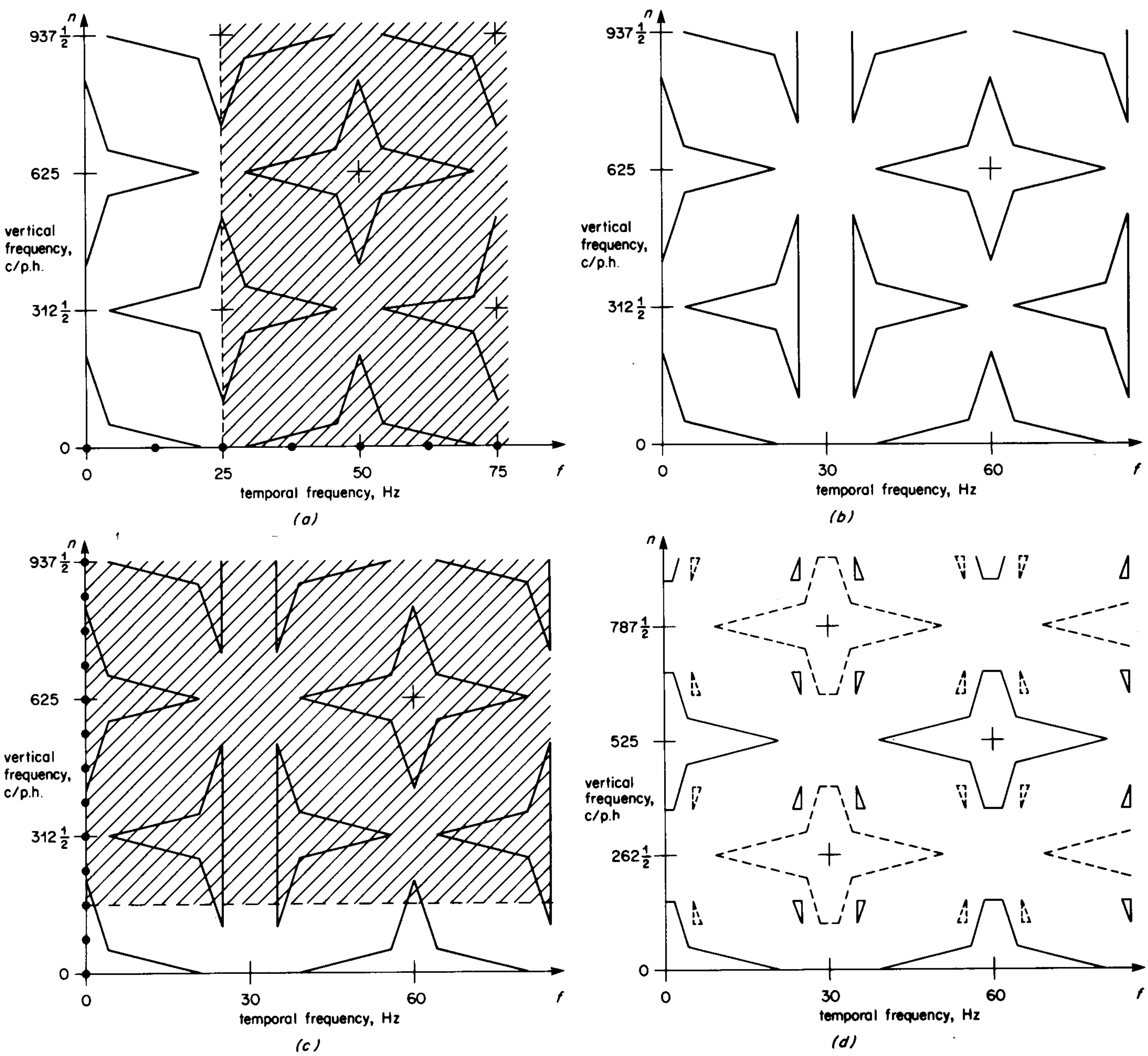


Fig. 35 – A spectral representation of temporal interpolation followed by vertical interpolation for interlaced scanning: (a) temporal low-pass filtering and (b) resampling to produce a sequential intermediate standard, followed by (c) vertical low-pass filtering and (d) resampling (the positions of spectra for interlaced output fields are shown dashed).

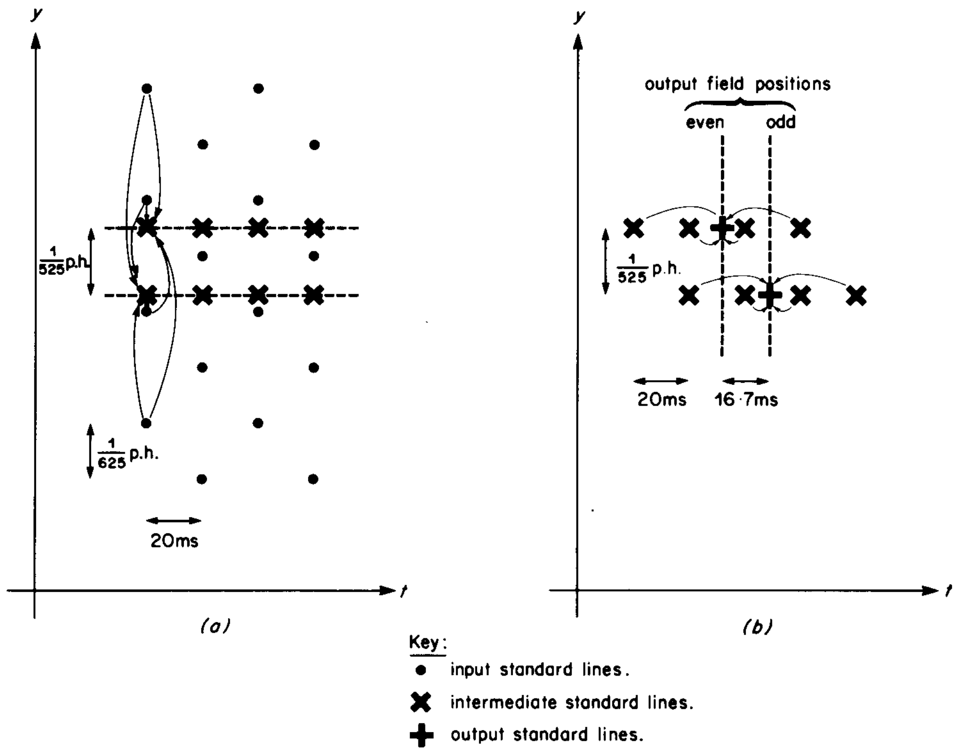


Fig. 36 – Vertical interpolation followed by temporal interpolation for interlaced scanning: (a) lines of a sequential intermediate standard are interpolated for new vertical positions, but at the same positions in time as the input fields; (b) lines from different fields of the intermediate standard are used to produce lines at new temporal positions appropriate for the even and odd positions on alternate output fields.

baseband information, but leaves the interlaced spectra virtually unsuppressed. Temporal resampling at the new field rate of 60 Hz produces the 625/60 sequential spectrum of Fig. 35 (b).

The vertical interpolator now requires a very abrupt cut-off at a low vertical frequency ($156\frac{1}{4}$ c/p.h.) in order to suppress the remnants of the interlaced components left by the temporal interpolator. This is shown in Fig. 35 (c). Vertical resampling at 525 lines per field would produce only the components shown as solid lines in Fig. 35 (d); however, when new lines are interpolated only at even line positions on one field and at odd line positions on the next, the extra components resulting from interlacing are introduced (shown dashed in Fig. 35 (d)).

The same performance can be obtained when vertical interpolation precedes temporal interpolation. In this case, lines from one input field are used to interpolate new lines at positions corresponding to the output standard line spacing, as shown in Fig. 36 (a). The 625/50 interlaced input is

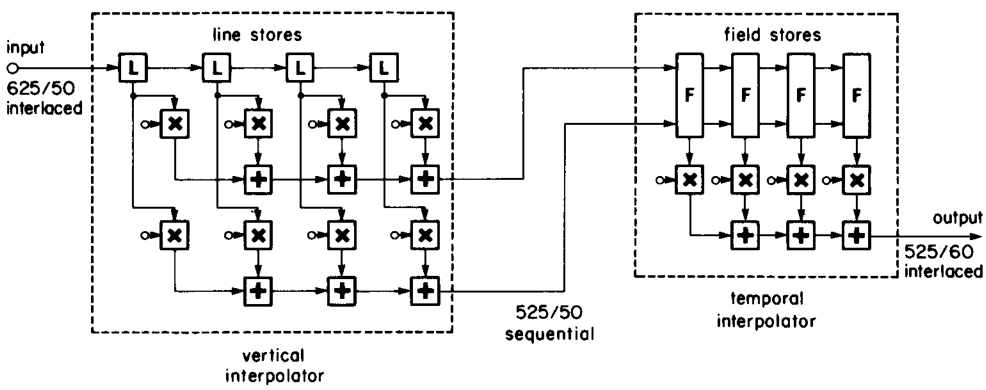


Fig. 37 – A converter arrangement for interlaced scanning in which vertical interpolation is followed by temporal interpolation. Note that the field stores each require twice the capacity of those in Fig. 34 as sequential fields are being stored.

therefore converted to a 525/50 sequential intermediate standard. The temporal interpolator then converts the intermediate standard into the output standard using sets of lines at the same vertical position from several fields (Fig. 36(b)); for an interlaced scan, lines are produced only at the odd and then the even positions on alternate output fields.

The block diagram of a converter using vertical, followed by temporal interpolation is shown in Fig. 37. Comparing this with Fig. 34, the same number of multipliers and adders are used in each case, but the storage requirements differ. Fig. 34 with its eight line stores appears the more complicated, but the field stores in Fig. 37 need approximately twice the capacity of those in Fig. 34 because sequentially scanned fields are being stored.

In the frequency domain, Fig. 38, the vertical interpolator has to reject both the vertical and interlaced components to produce a 525/50 sequential intermediate standard (Fig. 38 (b)). This is then converted to the interlaced 525/60 standard shown in Fig. 38 (d).

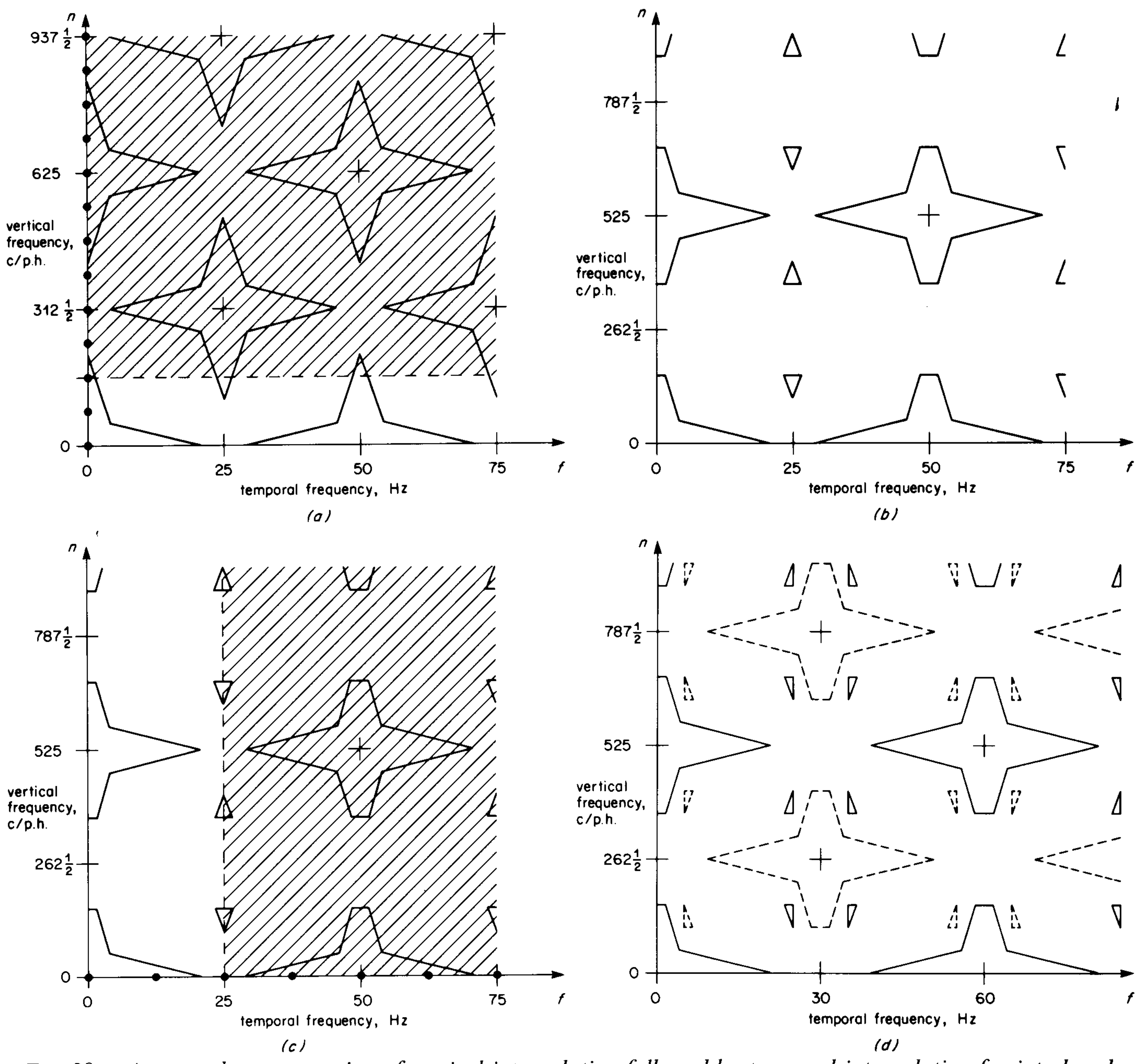


Fig. 38 – A spectral representation of vertical interpolation followed by temporal interpolation for interlaced scanning: (a) vertical low-pass filtering and (b) resampling to produce a sequential intermediate standard, followed by (c) temporal low-pass filtering and (d) resampling (the positions of the interlaced spectra are shown dashed).

As the overall effect of the temporal and vertical interpolation processes is the same whichever order is used, the smaller storage capacity required for Fig. 34 makes temporal followed by vertical interpolation preferable. In either case, the rectangular form of the overall frequency characteristic which, as explained earlier in this Section, results with any variables separable aperture is not a good approximation to Fig. 29, so that vertical resolution is significantly impaired and the original interlaced components are not fully suppressed.

If an intermediate standard with a normal line rate were to be used (instead of the double line rate, sequential scan produced by the first interpolation process), this would result in the generation of extra spectral components in either Fig. 35(b) or Fig. 38(b). The second one-dimensional interpolation process would be unable to suppress these extra components effectively, so that additional impairments would be introduced unnecessarily.

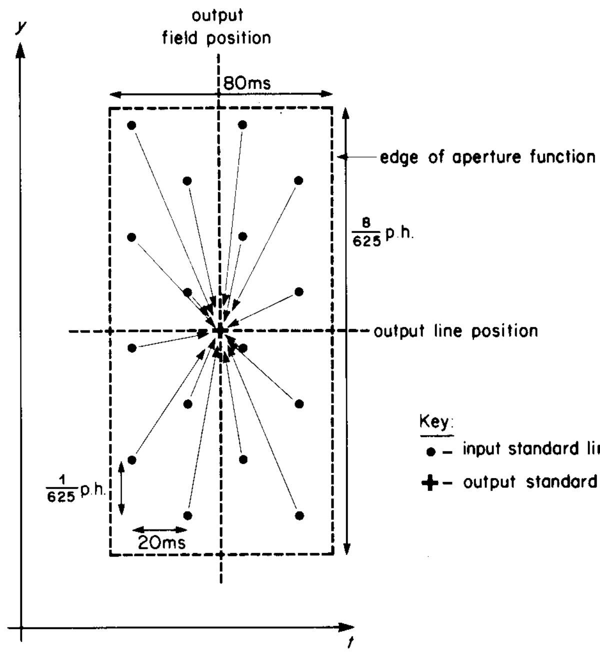


Fig. 39 – Combined vertical and temporal interpolation: each of the sixteen input lines falling within the aperture is weighted according to its individual temporal and vertical offsets from the required output line position.

3.2.2. Combined interpolation

If vertical–temporal interpolation is treated as a single process, the two-dimensional aperture functions and frequency characteristics are not restricted to variables-separable functions and, consequently, to rectangular frequency characteristics. It is possible, therefore, to obtain a close approximation to the triangular shape of frequency characteristic required for interlaced scanning.

The process of combined interpolation is illustrated in Fig. 39 which shows that all the input standard lines falling within the two-dimensional interpolation aperture contribute directly to one line of the output standard. Because the process is two-dimensional, the weighting coefficient applied to each line can take account of both its temporal and vertical offsets. Conceptually, the aperture function is placed over the pattern of the input lines with its origin at the required output line position; this identifies the aperture function value (weighting coefficient) appropriate for each of the input lines.

In the combined interpolator, the extent of the aperture function determines the amount of storage required. Fig. 40 shows an interpolator using four lines from each of four fields, which is the same as the effective aperture size given by the two one-dimensional interpolators in Figs. 34 and 37. The combined two-dimensional interpolator requires rather more line stores, multipliers and adders than the separate one-dimensional interpolators. However, the line stores, although shown in Fig. 40, can be omitted if multiple-output field

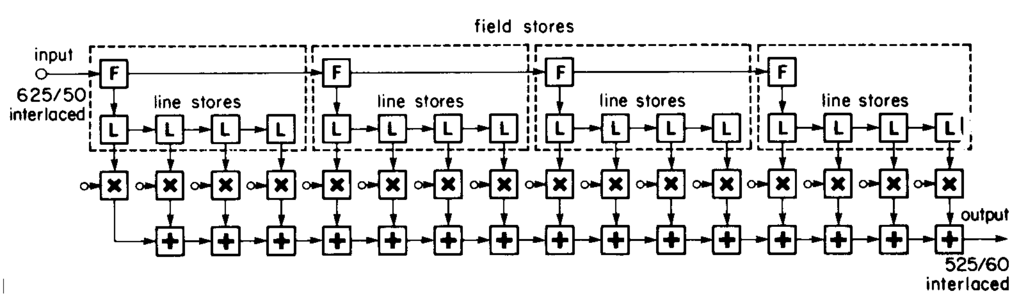


Fig. 40 – A converter arrangement for combined vertical and temporal interpolation. The line stores can be omitted if multiple output field stores are used, as indicated by the dashed outlines.

stores are used.* The complexity involved in the two methods is then broadly similar.

In the frequency domain, the two-dimensional frequency characteristic rejects all the repeated spectra at once, as shown in Fig. 41 (a). In this case, because it is non-variables-separable, the frequency characteristic can be specified independently at $78\frac{1}{8}$ c/p.h. and $12\frac{1}{2}$ Hz intervals all over the vertical–temporal spectrum. Compared with variables-separable functions, this provides much more opportunity to match the frequency characteristic of the interpolator to the shape of the scanned spectrum. In particular, the vertical resolution for still pictures (along the *n* axis) can be extended, relative to that shown in Figs 35 (c) and 38 (a), while still suppressing the main interlace components centred on $(25, 312\frac{1}{2})$. The repeated spectra are then reintroduced directly at the output standard positions, as shown in Fig. 41 (b).

It should be noted that although the particular shape of baseband spectrum used in the frequency domain diagrams is representative of most pictures, some components can extend well beyond this outline. Therefore, in all the methods of conversion described, substantial overlapping of the spectral components can sometimes occur.

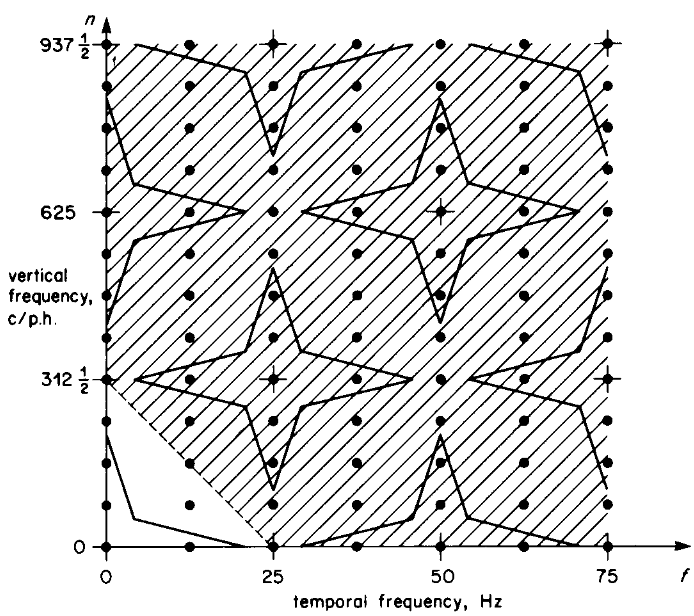
* The use of multiple-output field stores was devised by G. D. Roe and C. J. Dalton in BBC Engineering Designs Department; UK Patent Application No. 7901511.

3.3. Effects of inadequate interpolation

The sharp cut-off of the frequency characteristic shown in Fig. 29, although possibly ideal in theory, is not practicable and may not even be desirable. Realisable interpolation methods do not give the abrupt cut-off filtering characteristics and the neat separation of the spectral components indicated in the frequency domain diagrams of Section 3.2. This next Section describes the mechanisms causing particular forms of impairment which arise from non-ideal filtering, first in general terms. This is followed by a consideration of the impairments introduced by two previous converters, the performance characteristics of which are well known.

3.3.1. Forms of impairment

A filter with a gradual cut-off may both attenuate the wanted baseband components and leave some parts of the repeated spectra unsuppressed. Much of the signal energy is concentrated along the axes of the baseband spectrum and in corresponding areas of the repeated spectra. However, the centres of the repeated spectra are easy to suppress and the higher order spectra only lead to impairments when very simple aperture functions are used. Therefore, the main impairments result from the response of the filter characteristic in the regions labelled A to F in Fig.



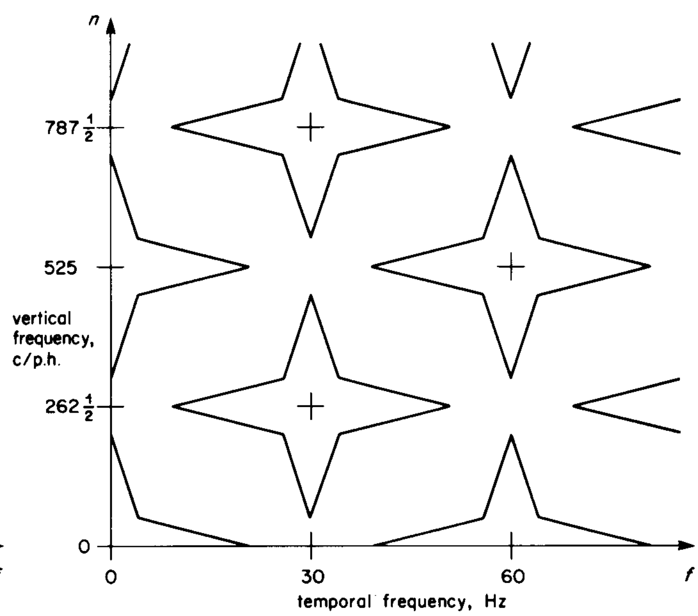


Fig. 41 – Spectra showing the effect of combined vertical and temporal interpolation for interlaced scanning: (a) all the repeated spectra are suppressed by the two-dimensional low-pass filtering action of the interpolator and (b) the resampling action produces new repeated spectra in both dimensions directly; with an aperture of four fields by four lines (Fig. 40) the frequency characteristic can be set independently at the positions shown in (a).

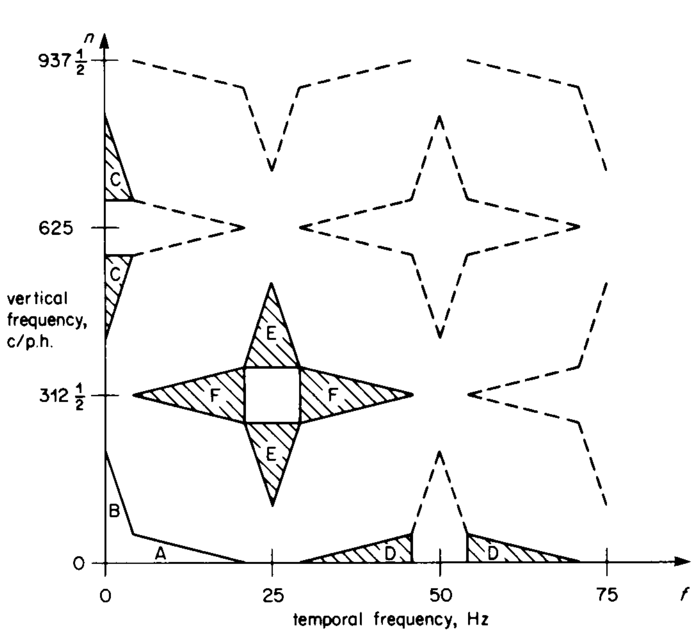


Fig. 42 – The low-pass filtered spectrum of interlaced scanned television signals: regions which produce the main impairments when converted to a new standard. Passband impairments: insufficient response in Region A causes blurring of movement and in Region B results in a loss of vertical detail. Stopband impairments: insufficient suppression of region C results in vertical aliasing, region D in movement judder, region E in flicker on vertical detail, while insufficient suppression of region F results in vertical modulation of moving detail.

42. Inadequate passband response in region A will result in blurring of movement whilst attenuation in region B will cause a loss of vertical detail. The other impairments, resulting from inadequate suppression of regions C, D, E and F, occur when these harmonic components are aliased by the resampling process to fall within the baseband spectrum.

The aliased spectra are offset from the origin by the difference between the old and new sampling frequencies. Thus, when resampled on the 525/60 standard, the components from region C, shown shaded in Fig. 42, are centred on 100 c/p.h., as shown in Fig. 43 (a). In addition, the corresponding components from $(0, -625)$ are shifted to $(0, -100)$ in Fig. 43 (a). The results for 525/60 to 625/50 conversion are substantially the same, although components from $(0, 525)$ and $(0, -525)$ are shifted to $(0, -100)$ and $(0, 100)$ respectively. The main components in region C are vertical frequencies from stationary pictures and these result in alias components which superimpose different vertical frequencies, offset by 100 Hz from the true frequencies. With high contrast signals, the non-linearity of the display tube produces a 100 c/p.h. beat frequency component between the true and aliased patterns. In pictures, this vertical aliasing appears mainly as 'knotting' on diagonal edges.

The corresponding effect for temporal frequencies is that components from region D, centred on $(50, 0)$ and $(-50, 0)$ appear as alias components centred on $(-10, 0)$ and $(10, 0)$ respectively in the 525/60 spectrum, as shown in Fig. 43 (b). For 525/60 to 625/50 conversion, the diagram is similar although the components at $(10, 0)$ and $(-10, 0)$ are derived from $(60, 0)$ and $(-60, 0)$ respectively. This temporal aliasing appears as judder on moving objects, caused by the alias components perturbing the true rate of movement.

In Fig. 42, frequencies along the 625 c/p.h. line and the 50 Hz line are easy to suppress. Consequently temporal components from the $(0, 625)$ spectrum and vertical components from the $(50, 0)$ spectrum usually cause no problems. However, both the vertical components (region E) and the temporal components (region F) of the interlaced spectrum centred on $(25, 312\frac{1}{2})$ can lead to individual aliasing problems.

For vertical frequencies, the shaded areas of Fig. 43 (c) show the positions occupied by the main alias components after resampling. In this case, the aliased spectra are offset vertically by 50 c/p.h. and temporally by 5 Hz so that stationary vertical frequencies result in vertical detail flashing at a 5 Hz rate. This effect is known as 5 Hz flicker and often appears as a repeated upward and downward displacement of the horizontal boundaries of objects.

For temporal frequencies, the region F components from Fig. 42 are moved to the 50 c/p.h. lines of the baseband spectrum as shown in Fig. 43 (d). Because of this, an object moving horizontally, which originally had no vertical detail, would have a 50 c/p.h. vertical pattern superimposed on its moving vertical edges. In its most severe form, this impairment is known as 'crankshaft' distortion because of its jagged, stepped appearance.

As shown in Figs. 43 (c) and (d), the interlaced spectra produce four separate alias components from the spectra at $(\pm 25, \pm 312\frac{1}{2})$ in the four quadrants. Again 525/60 to 625/50 conversion results in the same frequency offsets shown in Fig. 43, although the alias components are reversed in their positions. All the individual alias components shown in Fig. 43 are attenuated by the low-pass filtering effect of the interpolation process and

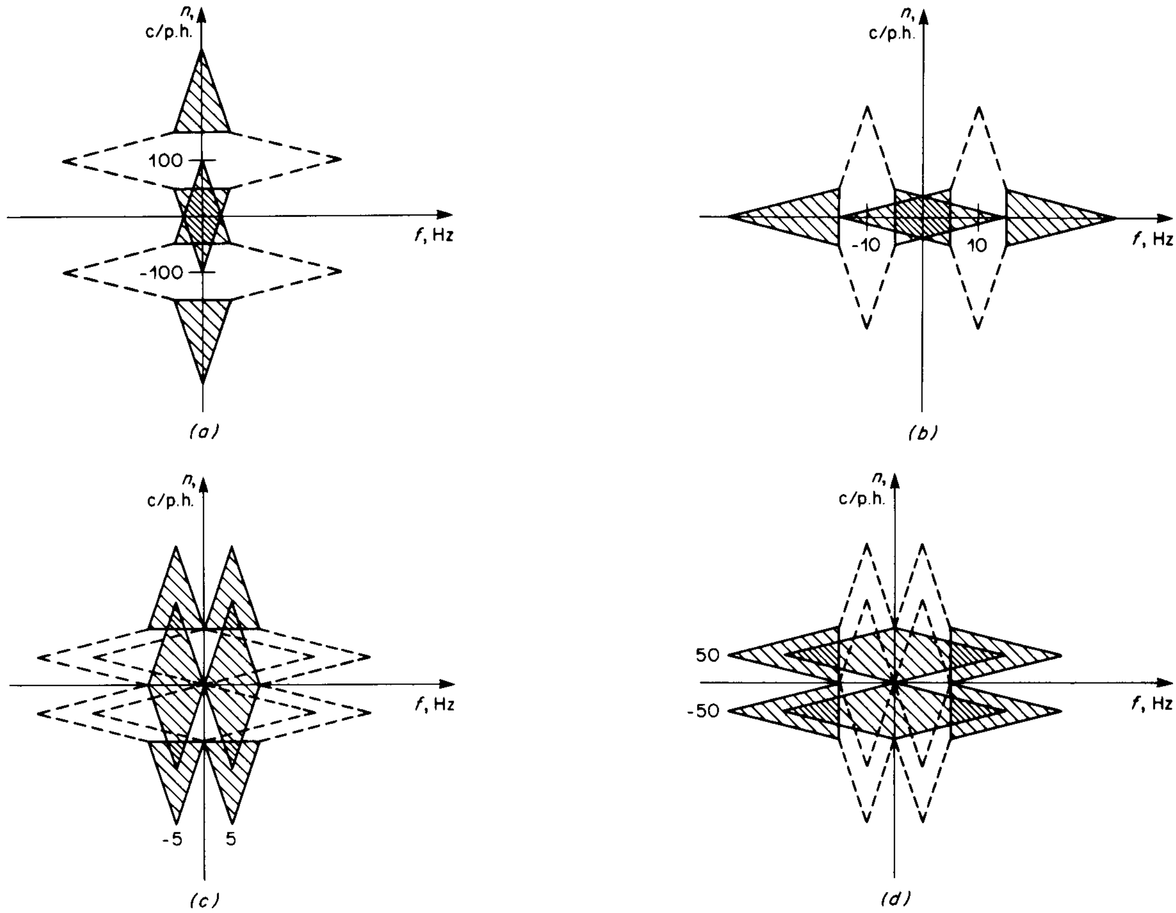


Fig. 43 – The effect of resampling the spectrum of Fig. 42 using the 525/60 standard: (a) components from region C are shifted to be centred on (0, 100) causing 100 c/p.h. vertical aliasing (in addition, similar components are shifted from (0, −625) to (0, −100)), (b) region D components are shifted to (±10, 0) causing 10 Hz movement judder, (c) region E components are shifted to (±5, ±50) causing 5 Hz flicker, and (d) region F components are shifted to (±5, ±50) causing 50 c/p.h. modulation of moving edges.

should be superimposed, together with the original baseband spectrum, to give the overall spectrum resulting from standards conversion. This spectrum is then repeated at all multiples of the output standard scanning rates.

As mentioned in Section 2.2.1, changing from a higher to a lower sampling rate can result in aliasing on the output standard. This could occur with 625 to 525 conversion in which the extra vertical detail of the 625-line system could cause 30 Hz interlace flicker on the output standard. Also, with 60 to 50 Hz field-rate conversion, the extra temporal components could result in the individual fields becoming more visible on movement. However, both these impairments are accepted as normal features of the output standard television system; so it is much less important for these to be suppressed than for the abnormal alias components shown in Fig. 43. Even so, it is relatively easy to suppress components that would cause interlace flicker without sacrificing much useful vertical resolution. In this respect, then, the converted pictures can be of higher quality than pictures directly scanned on the output standard.

3.3.2. Performance of previous converters

As a further illustration of the frequency domain method, the forms of impairment produced by two well known interpolation methods will be related to features of their frequency characteristics. The first interpolation system is that used in the analogue converters[1,2] developed by BBC Research Department and first used in June 1968. The second method is an approximation to that used in the IBA's DICE converters,[3,4] first used for 525/60 to 625/50 conversion in November 1972 and for the reverse direction in May 1975. Almost all of the broadcast pictures converted in the United Kingdom during the 1970s used one of these two methods.

The BBC analogue converters used quartz ultrasonic delay lines equivalent to two fields of storage and f.m. signal averagers for interpolation.

Thus, many impairments, such as spurious responses in the delay lines and slight gain variations between the lines, were the result of the analogue circuitry rather than the interpolation method. Also, the f.m. averagers were only able to produce relatively simple combinations of signals for interpolation.

However, the interpolation method is well defined and is equivalent to an interpolation aperture function* one quadrant of which is shown in Fig. 44. In terms of the time domain performance, this can be interpreted as a combination of two methods. If the required output line position is close to an input field, such as in Fig. 45 (a), that is, within $\pm\frac{1}{4}$ field period, then the two closest lines of that input field are combined with weights of $\frac{3}{4}$ for the closer of the two and $\frac{1}{4}$ for the other line. When the output position is outside the $\frac{1}{4}$ field period limit, such as in Fig. 45 (b), the nearest lines of the preceding and succeeding fields are averaged (one line from each). It should be noted that, although each of these methods alone would be variables-separable, the combination of the two, shown in Fig. 44, is non-variables-separable. The same interpolation method was used in both directions of conversion.

*The mode of operation of this converter was expressed in terms of an aperture function by J. O. Drewery.

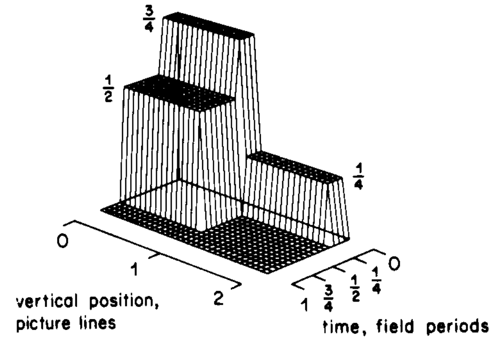


Fig. 44 – The two-dimensional aperture function corresponding to the interpolation method used in the BBC analogue converters. Only one quadrant is shown.

The frequency characteristic corresponding to this aperture function can be calculated by two-dimensional Fourier transformation. Fig. 46 (a) shows the characteristic for the 625/50 to 525/60 direction of conversion as a perspective view. However, the contour plot representation, Fig. 46 (b), showing the magnitude of the characteristic, is more useful for assessing the suppression of alias components. The centres of the repeated spectra, marked with crosses in Fig. 46 (b), are all well suppressed, but components corresponding to regions C, D and E of Fig. 42 all contain significant

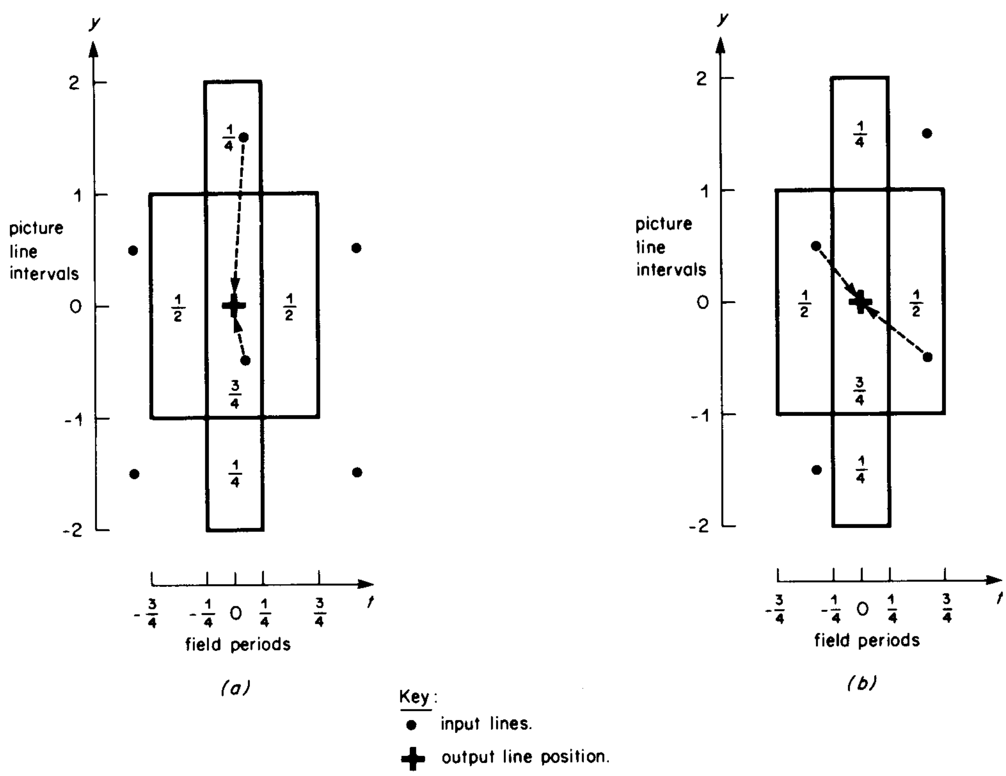


Fig. 45 – Interpolator action in the BBC analogue converters: (a) when the required output line position is within $\pm\frac{1}{4}$ field period of an input field position, two lines from that field are added with weights of $\frac{3}{4}$ for the nearest and $\frac{1}{4}$ for the next nearest; (b) when the required output line position is between two input fields, the two nearest lines are added, one from each field, with weights of $\frac{1}{2}$.

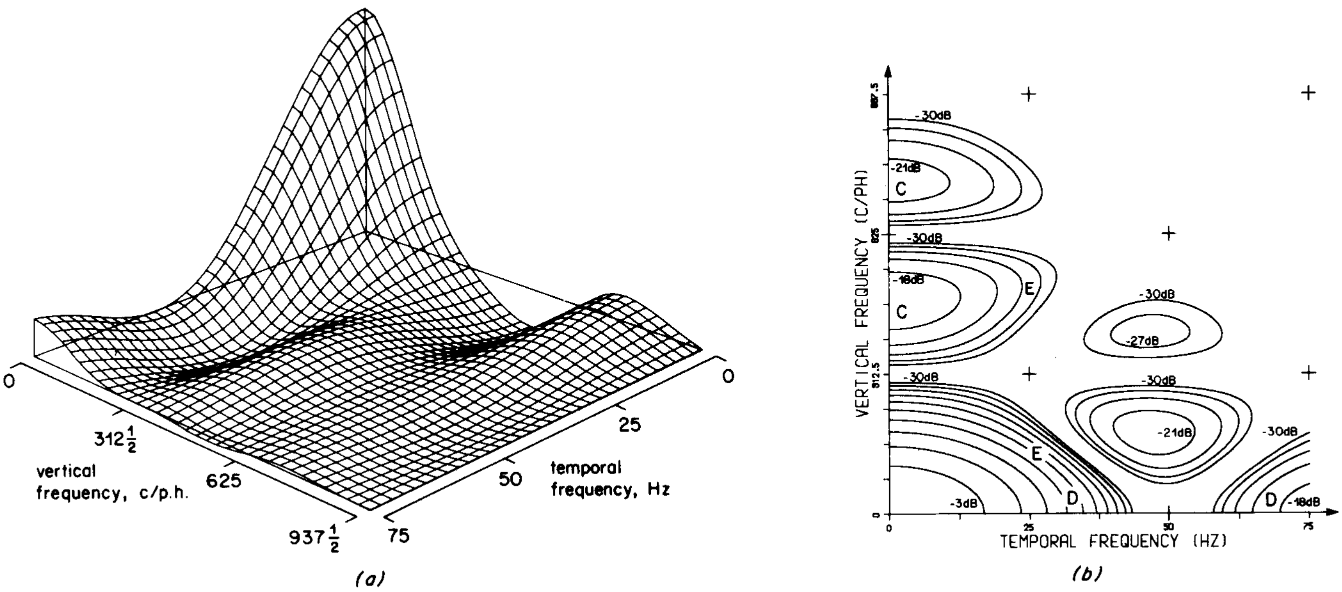


Fig. 46 – The vertical–temporal low-pass filter characteristic of the BBC analogue converters shown for 625/50 to 525/60 conversion: (a) as a perspective view and (b) as contours of equal attenuation; contours are spaced at 3 dB intervals down to –30 dB. The regions C, D, and E show the degree of attenuation provided for each of the main impairments marked in Fig. 42.

components which cause vertical aliasing, movement judder and 5 Hz flicker, respectively. As there is no response along the 312½ c/p.h. line, no aliasing corresponding to region F should be present. These forms of impairment agree well with the observed performance of this type of converter. The additional responses along the 50 Hz line lead to 10 Hz flicker on vertical detail. However, in pictures this would be difficult to distinguish from the higher amplitudes of 5 Hz flicker also produced on vertical detail by the region E components.

In marked contrast to this analogue conversion method, the interpolation of DICE uses digital storage and arithmetic. Because of this, the interpolation multipliers are much more sophisticated and give greater freedom for the design of aperture functions. However, because it was designed at a time when digital storage was still expensive, the capacity of the main store is only sufficient for two composite interlaced fields on the 525-line NTSC colour standard. This restriction, combined with the use of separate vertical and temporal interpolation, necessitates a significantly different mode of operation in the two directions of conversion.

For 625/50 to 525/60 conversion the arrangement shown in Fig. 47 (a) is used. Vertical (line) interpolation using five field-lines is performed first so that the resulting 525/50 interlaced signals can be stored. Following each store output there is a second vertical interpolator using three field-lines which converts the stored 525/50 interlaced signals to the positions of 525/50 sequential scanning. The combined effect of these two interpolations is, therefore, similar to that shown in Figs. 38 (a) and (b). This is followed by temporal (movement) interpolation using an approximation to a two-field linear aperture, resulting in spectra corresponding to those in Figs. 38 (c) and (d).

For the reverse direction of conversion, the 525/60 interlaced signals can be stored directly in the field stores, as shown in Fig. 47 (b). Again there are two vertical processes, but in this case these are placed one before and one after temporal interpolation. First, the three-line vertical interpolation produces what amounts to 625/60 sequential signals, in which the lines resulting from odd and even input fields are vertically coincident, but irregularly spaced. This allows temporal interpolation using an approximation to a two-field linear aperture function which produces 625/50 signals, although these still include the irregular line spacing. Finally, at the output, the regular, interlaced 625-line spacing is restored by the five-line vertical interpolator.

Specific details of the two vertical interpolation processes are not given.[3,4] Even so, a reasonable indication of performance can be obtained by assuming a single sinc function approximation extending over five field-lines for the combined effect of the two processes. This leads to an optimistic view of performance because it amounts to assuming that the three-line interpolator produces perfect results. Convolving this assumed form of vertical interpolation aperture

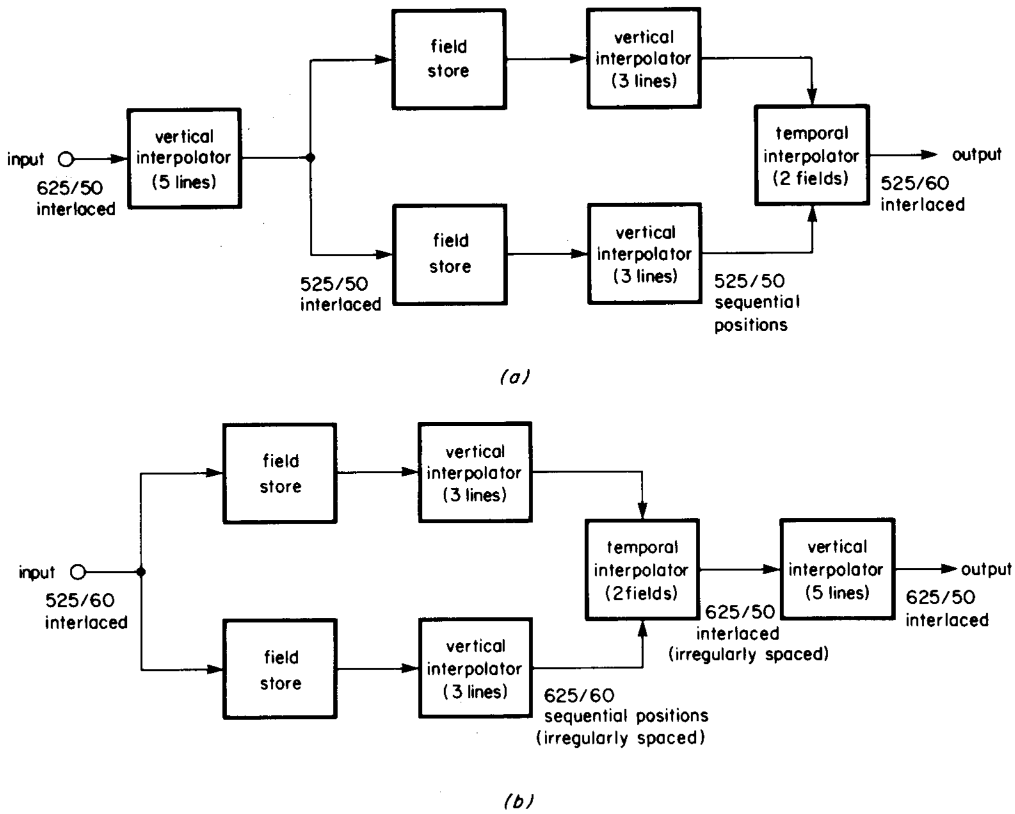


Fig. 47 – Block diagrams showing the sequence of interpolation processes in the IBA's DICE converter: (a) for 625/50 to 525/60 conversion and (b) for 525/60 to 625/50 conversion.

with the quantised two-field linear aperture used for temporal interpolation (625/50 to 525/60 direction) produces the two-dimensional aperture function shown in Fig. 48.

Fourier transformation of this aperture function produces the two-dimensional frequency characteristic shown in Fig. 49. Particularly for vertical frequencies, this is a much more effective low-pass filter characteristic than that shown in Fig. 46 for the BBC analogue converter. The contour plot representation, Fig. 49 (b), shows that all the components from region C in Fig. 42 are well suppressed, so that vertical aliasing is not a problem with this converter. In addition, there is greater suppression of the region D components at (75, 0), thus reducing movement judder, and less response in region E, reducing 5 Hz flicker. However, as expected for a converter with separate vertical and temporal interpolation, vertical resolution (region B) is very limited, with virtually no response beyond 156 c/p.h.

3.4. Aperture optimisation

Although the preceding analysis can identify the amount and form of the impairments resulting from an interpolation method, the degree of suppression required to render each impairment invisible can only be determined by experiment. Also when a balance has to be struck between impairments, this must take account of the visibility of each over a wide range of picture material. Because of this, a versatile experimental standards converter[7] was constructed to enable the actual performance of interpolation methods derived by

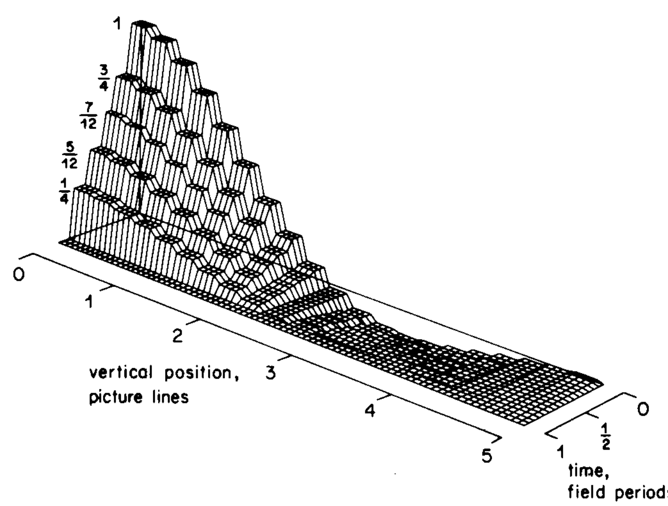


Fig. 48 – A two-dimensional interpolation aperture function approximating the interpolation methods used in the IBA's DICE converter for 625/50 to 525/60 conversion.

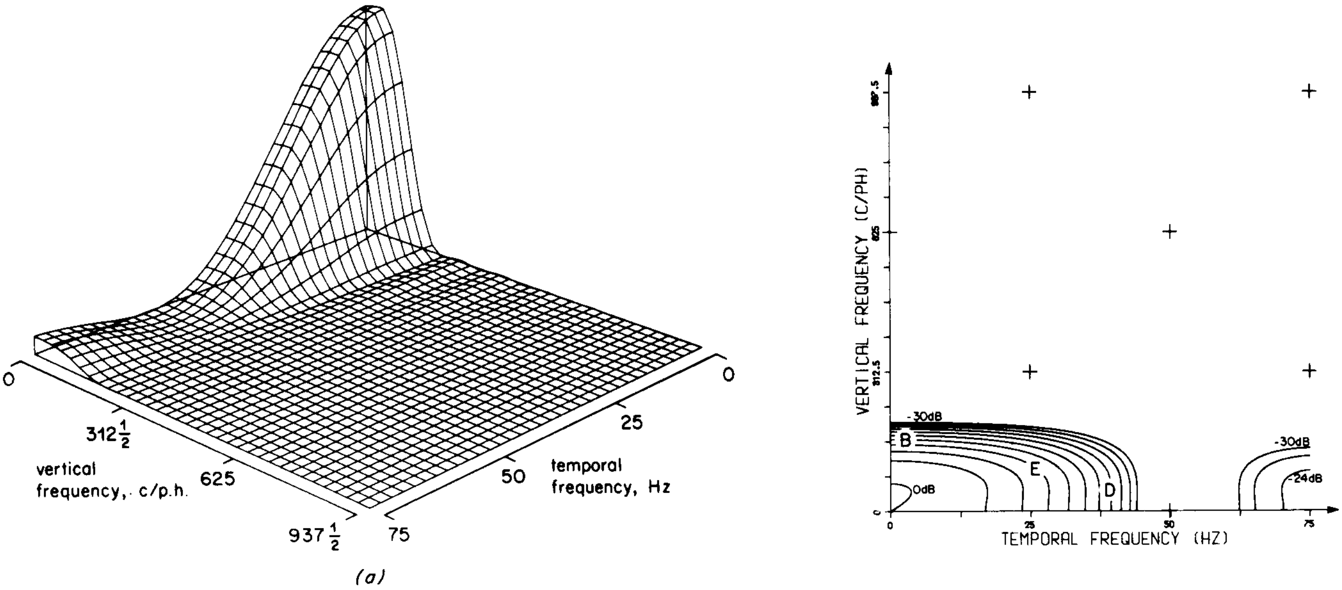


Fig. 49 – The vertical–temporal low-pass filter characteristic for the interpolation aperture function of Fig. 48: (a) shown as a perspective view and (b) as a contour plot. The responses in regions B, D and E show the main impairments to be vertical resolution loss, movement judder and 5 Hz flicker, respectively.

frequency domain analysis to be assessed. This allowed observed impairments to be related to particular features of the vertical–temporal spectrum, so that undesirable effects could be eliminated, where possible, by appropriate alterations to the frequency characteristic.

3.4.1. Aperture synthesis

With combined interpolation, the frequency characteristic can be set independently at fixed points spaced according to the aperture size, as shown in Fig. 41 (a). However, these fixed point values correspond to a repeated version of the aperture function, as described in Section 2.2.3 for the one-dimensional case. The frequency characteristic corresponding to the true aperture function is obtained by replacing each point value by a weighted two-dimensional sinc f. sinc n function and summing the individual contributions.

A frequency specification in which only one point has a non-zero value shows the form of the individual contributions. Setting (0, 0) to unity and zero values at all other specification points produces the frequency characteristic of Fig. 50, shown both as a perspective view and a contour plot. This shows that neighbouring set points are not interdependent because the function has zero values at these positions. However, a change at one set point would affect the function value between other points at the same temporal or vertical frequency over a wide range of frequencies. The aperture function corresponding to this frequency characteristic consists of a block pulse covering an area of four field periods by four field-lines.

For interlaced scanning, it is necessary to suppress frequency components beyond a line joining $(0, 312\frac{1}{2})$ and (25, 0), while minimising passband attenuation.

TABLE 2

vertical frequency (c/p.h.)	$312\frac{1}{2}$	0	0	0	0	0
	$234\frac{3}{8}$	1	0	0	0	0
	$156\frac{1}{4}$	1	0	0	0	0
	$78\frac{1}{8}$	1	1	0	0	0
	0	1	1	0	0	0
		0	$12\frac{1}{2}$	25	$37\frac{1}{2}$	50
		temporal frequency (Hz)				

The point values specified in Table 2 appear consistent with this, but produce the frequency characteristic of Fig. 51. The many overshoots in the characteristic result because the widely differing values at adjacent points in the specification make the passband to stopband transition too abrupt. The high level of stopband ripple makes the performance inadequate.

To obtain a smooth frequency characteristic, relatively free from overshoots, it is necessary to broaden the transition band. Even small encroachments into the stopband result in increased aliasing, so a more gradual cut-off has to be obtained

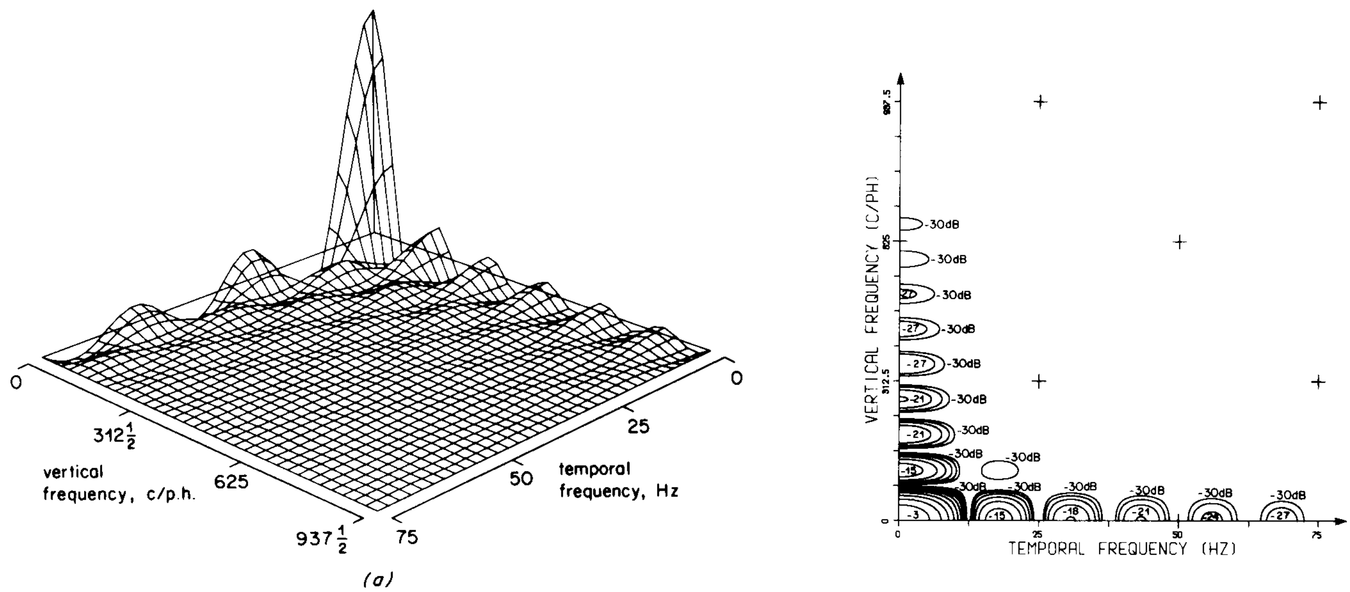


Fig. 50 – The vertical–temporal frequency characteristic resulting from a single non-zero fixed-point value for a combined two-dimensional interpolator with an aperture size of four fields by four field-lines (eight picture-lines): (a) perspective view and (b) contour plot.

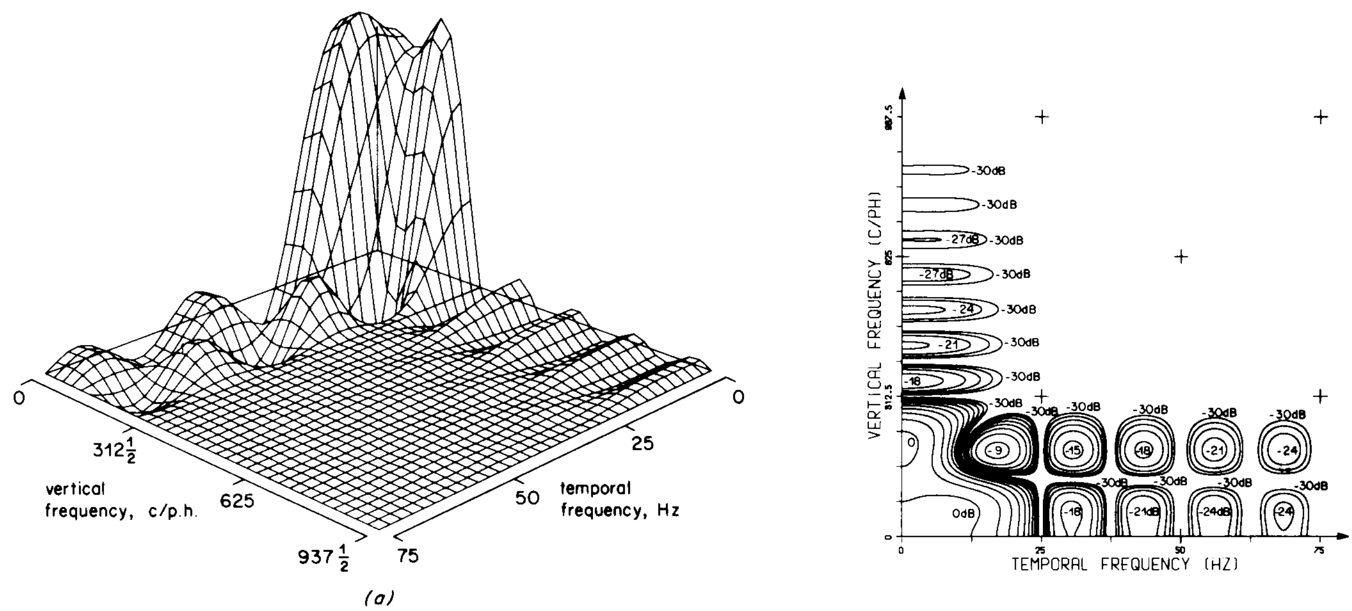


Fig. 51 – The vertical–temporal frequency characteristic resulting from the fixed-point values shown in Table 2: (a) perspective view and (b) contour plot.

primarily at the expense of passband performance. In addition, because the spacing of the temporal frequency specification points is wider, the cut-off in the temporal frequency direction will necessarily be more gradual than that for vertical frequencies. This, in turn, tends to restrict the vertical resolution because, if large values were set at high vertical frequencies and the triangular shape were maintained, this would cause too abrupt a cut-off in the temporal frequency direction.

TABLE 3

vertical frequency (c/p.h.)					
312½	0	0	0	0	0
234⅜	0.25	0.05	0	0	0
156¼	0.75	0.35	0	0	0
78⅛	1.0	0.7	0.1	0	0
0	1.0	0.8	0.2	0	0
temporal frequency (Hz)	0	12½	25	37½	50

Altering the frequency specification in accordance with these factors to the values of Table 3 produces the smooth, approximately triangular frequency characteristic shown in Fig. 52.

Having derived an apparently satisfactory frequency characteristic, the corresponding two-dimensional aperture function can be calculated as a summation of $\cos t.\ \cos y$ terms, weighted by the values in Table 3. Such a function was shown in its one-dimensional form in Section 2.2.3. This periodic function is then truncated to four field periods and eight picture-line intervals (four field-line intervals) to produce the aperture function. The first quadrant of the aperture function resulting from the set values in Table 3 is shown in Fig. 53.

3.4.2. Aperture quantisation in the experimental converter

For use in a digital converter, the aperture function shown in Fig. 53 must be quantised in time, vertical position and amplitude. As the aperture function has four-quadrant symmetry, only one quadrant need be stored. In the experimental converter, the aperture function was stored using 8 values per field period and 16 values per picture line interval. For an aperture the size of Fig. 53, therefore, a total of 1024 coefficients would be required to store one quadrant, consisting of 16 values in the temporal direction at each of 64 positions vertically.

Positional quantisation of the aperture function, in which N values are stored per sample period, repeats the baseband frequency characteristic at N times the sample rate (Section 2.2.4). So, with 8 values per field period and 16 values per picture line interval, extra responses in the characteristic are centred on harmonics of 400 Hz (8 times 50 Hz) in the temporal direction and 10,000 c/p.h. (16 times 625 c/p.h.) in the vertical direction. From Table 1, the minimum attenuation of alias components from these regions is at least 24 dB temporally ($N = 8$) and 30 dB vertically ($N = 16$). The actual attenuation values will be greater than this because the frequency characteristic of the continuous aperture function is not flat to the half sampling frequencies. For example, the characteristic shown in Fig. 52 has an attenuation of 14 dB at 25 Hz; so alias components due to aperture quantisation would be attenuated by a total of 38 dB in this case.

When only one quadrant is stored, only four of the sixteen coefficients (Fig. 39) can be read directly. The remaining coefficients are found by reflecting the line positions in other quadrants temporally and vertically about the aperture centre so that they fall into the first quadrant. In practice, when reading the coefficients from a memory, this is accomplished by complementing the address bits of the temporal component or of the vertical component, or of both together.

Each of the 1024 aperture values selected for storage must also be quantised in amplitude. As aperture functions generally include some negative values, an eight-bit representation consisting of seven amplitude bits and a sign bit is convenient.

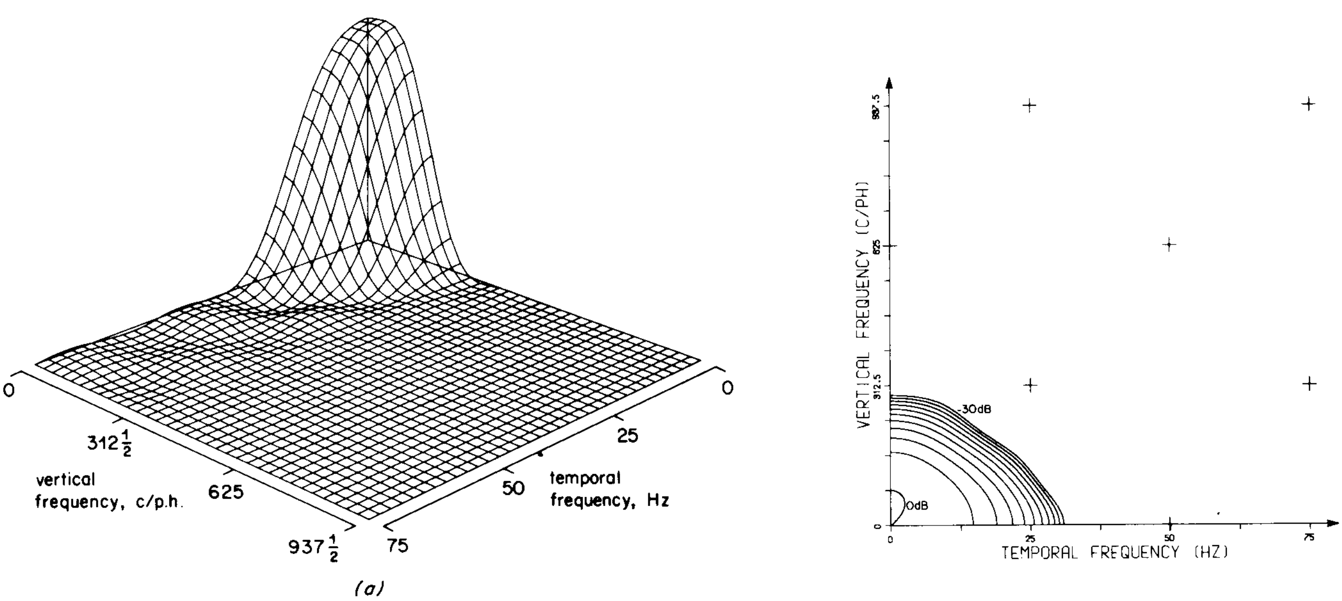


Fig. 52 – The vertical–temporal frequency characteristic resulting from the fixed-point values shown in Table 3, chosen to obtain a smooth response: (a) perspective view and (b) contour plot.

For all useful aperture functions the coefficient values are less than unity, so that the amplitude of each can be expressed as a whole number of 128ths.

As shown in Fig. 39, the coefficients are used in sets of 16, of which there are 64 different sets. Before amplitude quantisation, each set of coefficients sums to unity. However, when the coefficients are rounded individually to the nearest level, the sum may no longer be unity. This could cause brightness variations when changing from one set of coefficients to another. To avoid this the coefficients must be quantised in sets by a method which ensures that each set maintains its unity sum, even though this may increase the quantisation error of individual coefficients.

A suitable method is illustrated in Table 4. The first column shows, before quantisation, sixteen examples of aperture values to be applied as a set to sixteen input lines. The values shown have been multiplied by 128 so that after quantisation an integer value will remain; this indicates the coefficient value as a number of 128ths.

TABLE 4

Unquantised Values	Quantised Individually	Rounded Down	Quantisation Error	Quantised as a set
93.990	94	93	0.990*	94
1.364	1	1	0.364	1
0.153	0	0	0.153	0
0.000	0	0	0.000	0
14.949	15	14	0.949*	15
12.708	13	12	0.708*	13
−1.761	−2	−2	0.239	−2
−1.431	−1	−2	0.569*	−1
9.356	9	9	0.356	9
7.516	8	7	0.516*	8
−1.373	−1	−2	0.627*	−1
−1.073	−1	−2	0.927*	−1
−1.498	−1	−2	0.502	−2
−2.707	−3	−3	0.293	−3
−2.198	−2	−3	0.802*	−2
0.005	0	0	0.005	0
128.000	129	120	8.000	128

The values in the first column, therefore, sum to 128. Rounding each coefficient to the nearest quantised value produces the results in column 2 which sum to 129. Using this set of coefficients would produce a d.c. gain slightly greater than unity (other sets might produce gains both greater and less than this). Therefore, as a first step in the quantisation procedure all the coefficient values are rounded down as shown in column 3. The sum of these values is less than or equal to 128, in this case, 120. The errors resulting from rounding down are shown in column 4. To obtain a sum of 128 rather than 120, eight values must be rounded up instead of down. Selecting the eight coefficients with the largest errors from rounding down (marked with asterisks in Table 4) minimises the errors overall and produces the quantised set of coefficients shown in column 5.

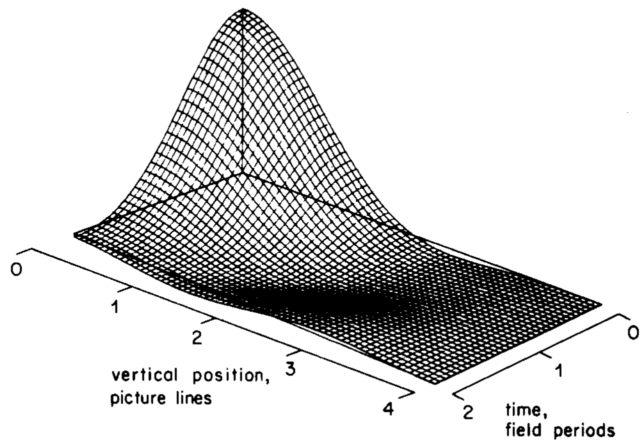


Fig. 53 – One quadrant of the two-dimensional aperture function corresponding to the fixed values of Table 3.

The frequency characteristic corresponding to the quantised aperture values, derived as described above, can be calculated by summing the Fourier transforms of individual block pulse elements (this was described for a one-dimensional aperture function in Section 2.2.4). The transform of the two-dimensional elements with amplitude A shown in Fig. 54 is given by:

$$G(f, n) = 4AT\cos 2\pi t_0 f \, . \, \mathrm{sinc}\, Tf \, . \, Y\cos 2\pi y_0 n \, . \, \mathrm{sinc}\, Yn$$

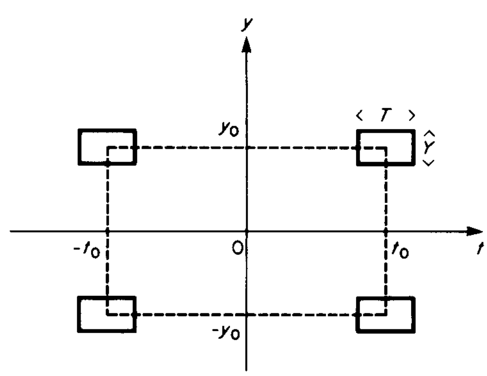


Fig. 54 – Symmetrical pairs of block pulses representing a generalised element of a quantised two-dimensional aperture function. Each block has an amplitude A.

The accuracy of quantisation used in the experimental converter is such that the frequency characteristic of the quantised function is not discernibly different from that shown in Fig. 52.

Use of the experimental converter to assess the performance of interpolation methods allowed the aperture functions to be further optimised. In particular, the main types of impairment described in Section 3.3.1 could be altered by adjustments to the corresponding fixed point values in the frequency specification. The results of further optimisation are described in Section 5.

4. Interpolation of decoded colour signals

In a digital standards converter, it is convenient to interpolate using separate component luminance and colour difference signals, such as *Y*, *U* and *V* or *Y*, *I* and *Q*, although these are often processed as a single time-multiplexed data stream. In most cases, these component signals are derived from encoded colour signals, such as PAL or NTSC, by an appropriate colour decoder. Because the decoding process is imperfect, some residual subcarrier will remain in the decoded luminance signals as cross-luminance. Also, some luminance signals will be demodulated to produce cross-colour. The form and extent of these crosstalk signals depends on the methods of filtering used to separate the luminance and chrominance signals in the decoder.[8]

The colour subcarrier frequencies used in the PAL and NTSC systems were chosen to minimise the visibility of the subcarrier dot pattern produced on monochrome receivers. Accordingly, in each case, the subcarrier has a high vertical frequency component to ensure that there is a phase change from line to line and a high temporal frequency component to produce a phase change from picture to picture. Because of this, the crosstalk components are centred on parts of the spectrum well separated from the main true signal components. Therefore, in a standards converter, the normal vertical–temporal low-pass filtering action of the interpolator will provide some attenuation of the crosstalk components left by the decoder. By taking account of the spectral positions of the crosstalk components when designing the interpolator, it is possible in some cases to improve this filtering action significantly.

4.1. Colour subcarrier in the vertical–temporal spectrum

The PAL colour subcarrier frequency of 4.43361875 MHz used in the 625/50 scanning system is generated as a one-dimensional signal. However, it can be interpreted as a scanned version of a three-dimensional signal, having components of horizontal, vertical and temporal frequency (*m*, *n* and *f*).[11] As a scanned signal, it is repeated around harmonics of the scanning rates and therefore appears at many positions in the *m*, *n*, *f* spectrum. When projected onto the *n*–*f* plane, the subcarrier positions are as shown in Fig. 55 (a).

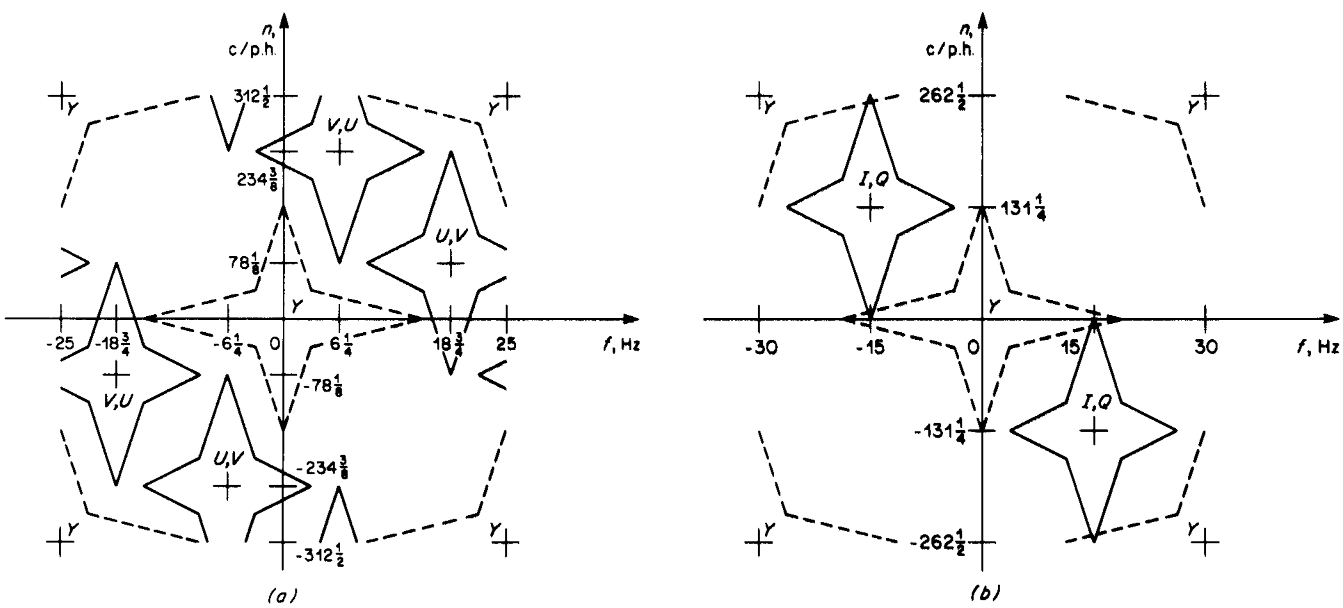


Fig. 55 – Positions of the main chrominance and luminance components in the vertical–temporal spectrum of encoded colour signals: (a) for 625/50 PAL and (b) for 525/60 NTSC. Luminance (Y) regions are marked with a dashed outline.

The positions of the U and V subcarriers are interchanged for positive and negative values of m. Thus, the U subcarrier is located at $(18\frac{3}{4}, 78\frac{1}{8})$, $(-6\frac{1}{4}, -234\frac{3}{8})$, etc. for positive values of m, and at $(6\frac{1}{4}, 234\frac{3}{8})$, $(-18\frac{3}{4}, -78\frac{1}{8})$, etc. for negative values of m. As well as being orthogonally phased, the V subcarrier has offsets of $-12\frac{1}{2}$ Hz and $156\frac{1}{4}$ c/p.h. from the U subcarrier position due to the modulating action of the V-axis switch. The positions shown in Fig. 55 (a) minimise the visibility of the subcarriers by giving the highest combinations of temporal and vertical frequency consistent with the offset resulting from the V-axis switch.

The colour difference signals produced from the image contain vertical and temporal frequency components similar to those for luminance described in Section 3.1. In the PAL encoded signal, these appear as sidebands extending from the colour subcarrier frequencies, Fig. 55 (a). For vertical frequencies, the sidebands extend from the subcarrier positions along lines parallel to the n axis, while for temporal frequencies, as might be produced by a coloured object moving horizontally, the sidebands extend parallel to the f axis. Although filtered by the effect of the television camera as described in Section 3.1, the modulation is not band-limited sufficiently to avoid aliasing; this will result for vertical modulating frequencies exceeding $312\frac{1}{2}$ c/p.h. or for temporal modulating frequencies exceeding 25 Hz. However, the colour difference signals usually have much lower horizontal bandwidth (approximately 1 MHz) than luminance signals. Because of this, horizontal movement produces much less high temporal frequency energy in the colour difference signals, so that temporal aliasing of chrominance is less likely to occur. For this reason the extent of the temporal chrominance sidebands shown in the diagram is less than that shown for luminance.

For 525/60 NTSC signals, the I and Q subcarriers are only separated by their orthogonal phase relationship. Therefore, the subcarriers are both located at the same positions in the n–f spectrum, as shown in Fig. 55 (b). As for PAL, this location combines the highest possible values of temporal and vertical frequency to give the minimum visibility of subcarrier. Because there is no offset to accommodate between the subcarriers, NTSC subcarriers are less visible than the equivalent PAL subcarriers for the same scanning standard. Vertical detail and horizontal movement result in modulating sidebands similar to those described for PAL.

4.2. Luminance filtering

The conventional method of suppressing the main subcarrier components in the luminance channel of a colour decoder is to use a notch filter, rejecting all horizontal frequencies at and around the subcarrier frequency. This sacrifices some horizontal resolution and fails to suppress the sidebands resulting from horizontal detail in the colour difference signals. In a standards converter, therefore, it is preferable to use a vertical filter based on line delays, (often referred to as a comb filter in this context) as this allows the full horizontal resolution to be preserved. This is particularly important for 525/60 to 625/50 conversion in which the 4.2 MHz resolution of the 525/60

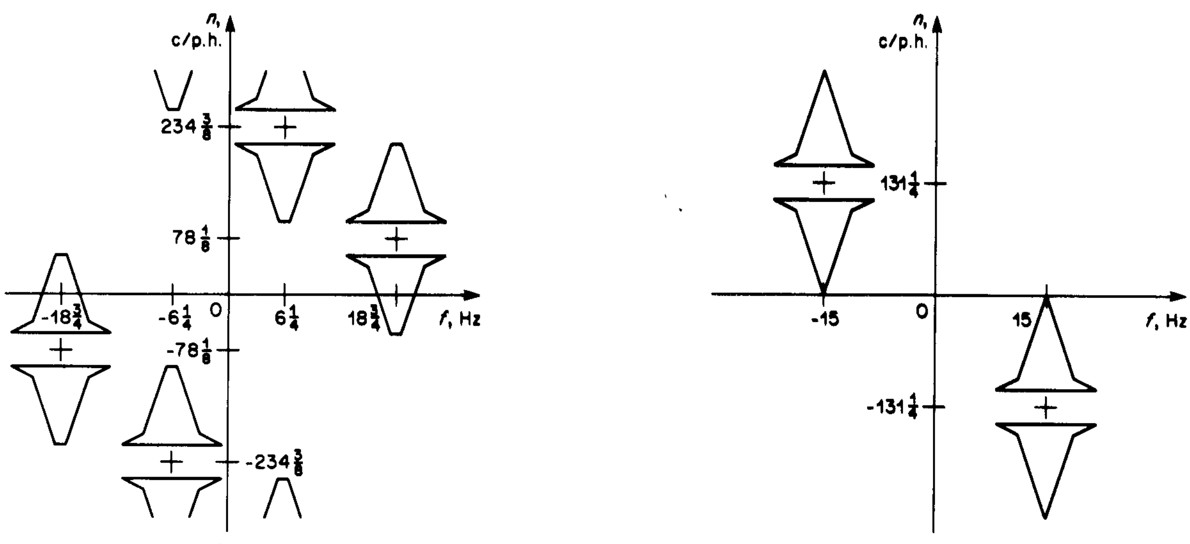


Fig. 56 – Suppression of the main subcarrier components (cross-luminance) in the vertical–temporal spectrum by using vertical filters (comb filters based on line delays): (a) for 625/50 PAL and (b) 525/60 NTSC.

standard is already significantly lower than the 5.5 MHz bandwidth of 625/50 PAL (System I).

The vertical filter rejects frequencies corresponding to the vertical component of subcarrier frequency, that is, $n = 78\frac{1}{8}$ and $234\frac{3}{8}$ c/p.h. in Fig. 56(a) or $n = 131\frac{1}{4}$ c/p.h. in Fig. 56(b), but sidebands resulting from vertical chrominance detail remain. Typical filter characteristics for PAL and NTSC are shown in Fig. 57. It should be noted that the effect of the vertical filters is limited to the high horizontal frequencies so that full vertical resolution is retained at low horizontal frequencies.

Besides residual subcarrier (cross-luminance), some decoders (those incorporating PAL modifiers or equivalent operations to cancel the effect of the V-axis switch) introduce luminance aliasing centred on twice subcarrier frequency. In the n–f plane, the positions nearest to the origin for twice subcarrier frequency are $(12\frac{1}{2}, -156\frac{1}{4})$ and $(-12\frac{1}{2}, 156\frac{1}{4})$; alias components extend from these centre frequencies as sidebands, as shown in Fig. 58(a). However, for some methods of decoding,[8] the luminance alias is filtered vertically so that the main alias components, resulting when vertical luminance detail is present, are located either side of twice subcarrier frequency, as shown in Fig. 58(b).

For 625/50 to 525/60 conversion, the filtering action of the interpolator aims to reject components beyond a line joining (25, 0) and $(0, 312\frac{1}{2})$, as shown in Fig. 59(a). Since the U and V subcarriers both lie on this line, the normal action of the interpolator will suppress the subcarrier centre frequencies. Also, because in practice the interpolator will have a gradual cut-off, the vertical and temporal chrominance sidebands will be somewhat attenuated. However, using the interpolator alone, it is not possible to provide complete

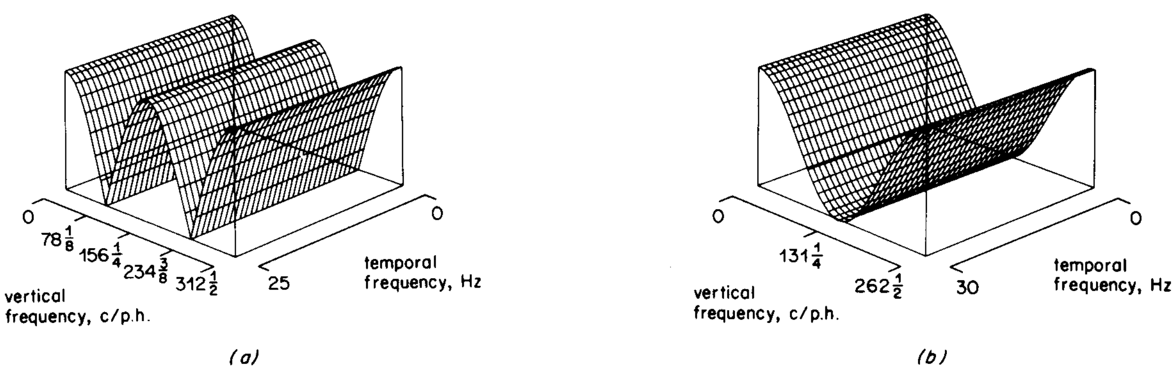


Fig. 57 – The vertical–temporal amplitude–frequency characteristics of vertical filters typical of those used for cross-luminance suppression: (a) for 625/50 PAL and (b) for 525/60 NTSC.

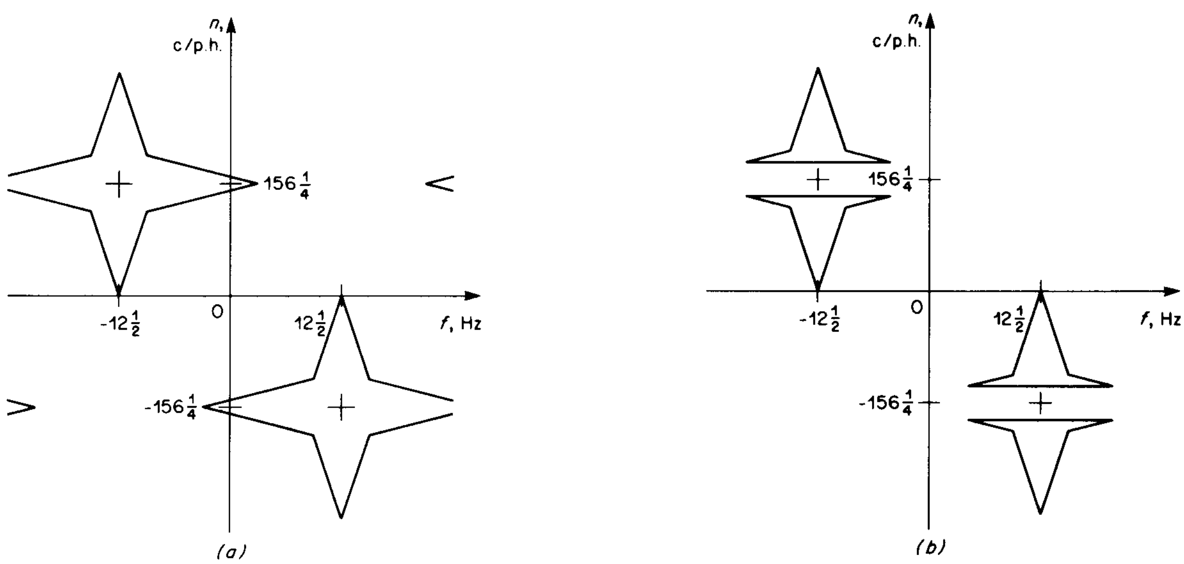


Fig. 58 – Positions in the vertical–temporal spectrum of the luminance alias components which result with PAL signals when decoders incorporating modifiers are used: (a) unfiltered alias components and (b) suppression of the alias component centres by vertical filtering.

suppression of the subcarrier components without impairing the wanted baseband luminance signals severely. Therefore, the optimum performance is obtained by using the vertical filter response, Fig. 57 (a), with an interpolator characteristic optimised primarily for the best suppression of alias components caused by scanning. Suppression of the PAL subcarrier is also aided by the 4.43 MHz subcarrier being outside the 4.2 MHz nominal luminance bandwidth of System M television. Similarly, using the interpolator, it is not possible to suppress luminance aliasing, resulting from the use of PAL modifiers, completely (Fig. 58). Therefore, it is preferable to select a decoder which does not introduce this impairment.

When the input standard is 525/60 interlaced, the interpolator is designed to reject components beyond a line joining (30, 0) and (0, $-262\frac{1}{2}$), Fig. 59 (b). As the I and Q subcarriers lie on this line, the main chrominance components are rejected as for PAL. However, the NTSC subcarrier is more difficult to suppress adequately because of its low horizontal frequency of approximately 3.58 MHz. The combined effect of the vertical filter response of Fig. 57 (b) and the normal interpolator characteristic provides good suppression of subcarrier except when the picture contains moving coloured vertical frequency detail. Then the subcarrier can become a stationary line-repetitive pattern, which is not attenuated. An example of the circumstances under which this can occur is in areas of crowd forming the background to sporting events.

4.3. Chrominance demodulation in the vertical–temporal spectrum

Chrominance demodulation consists of multiplying the modulated chrominance signals by subcarrier frequency sine and cosine waves.[8] Accordingly, in the frequency domain, the modulated chrominance spectrum is convolved with an impulse function at each subcarrier position. In the n–f plane, this shifts the U, I and Q components to be centred on the origin, as shown in Fig. 60. The spectrum for demodulated V signals is similar to Fig. 60 (a) with both the U and V positions, and the cross-colour $+m$ and $-m$ positions interchanged. Any luminance components entering the demodulators to produce cross-colour are similarly shifted in frequency to be centred on the positions previously occupied by the subcarriers. Thus, the cross-colour components $Y'(\pm m)$ are centred on $(6\frac{1}{4}, 234\frac{3}{8})$ and $(18\frac{3}{4}, 78\frac{1}{8})$ for PAL and $(-15, 131\frac{1}{4})$ for NTSC. The demodulation process also shifts chrominance information to twice subcarrier frequency but these components are removed by the horizontal low-pass filters of the demodulators.[8]

If the orthogonal phase relationship of the subcarriers has been maintained, there will be no crosstalk between the two chrominance channels. However, if there is differential phase distortion or if the alignment of either the coder or decoder is inaccurate, then some crosstalk will occur. In the PAL system, V components entering the U channel

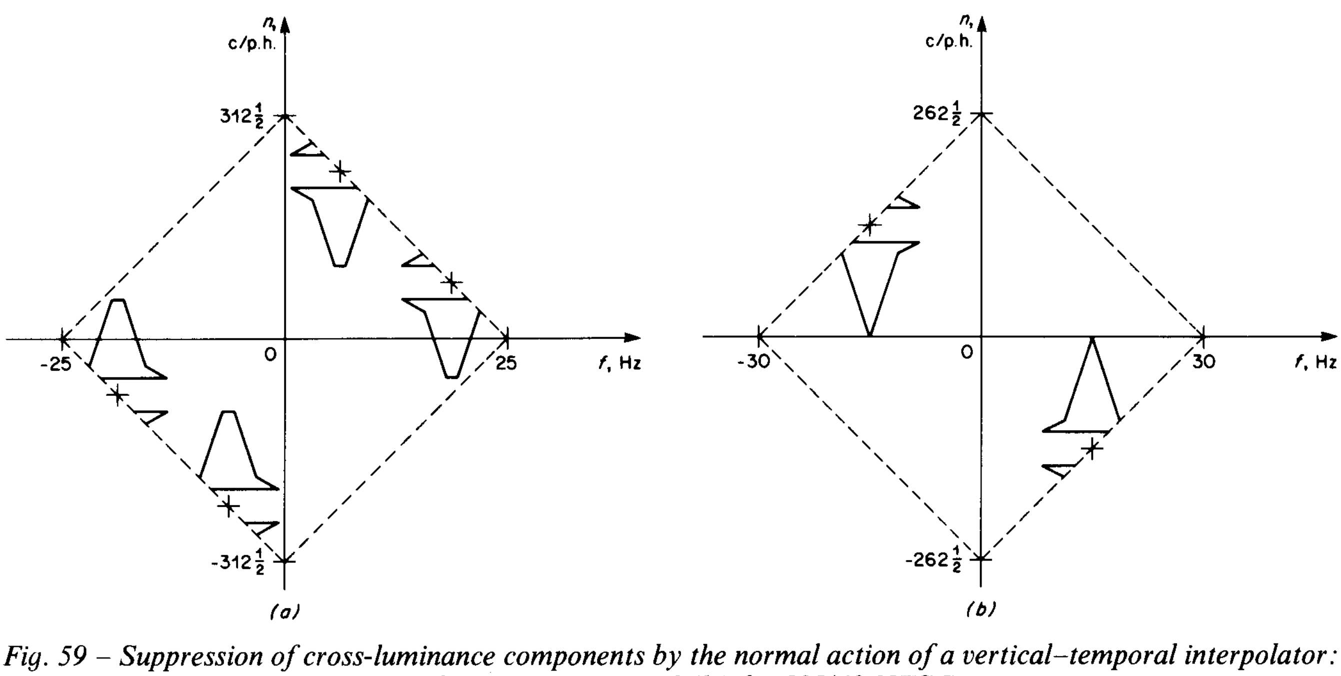


Fig. 59 – Suppression of cross-luminance components by the normal action of a vertical–temporal interpolator: (a) for 625/50 PAL and (b) for 525/60 NTSC.

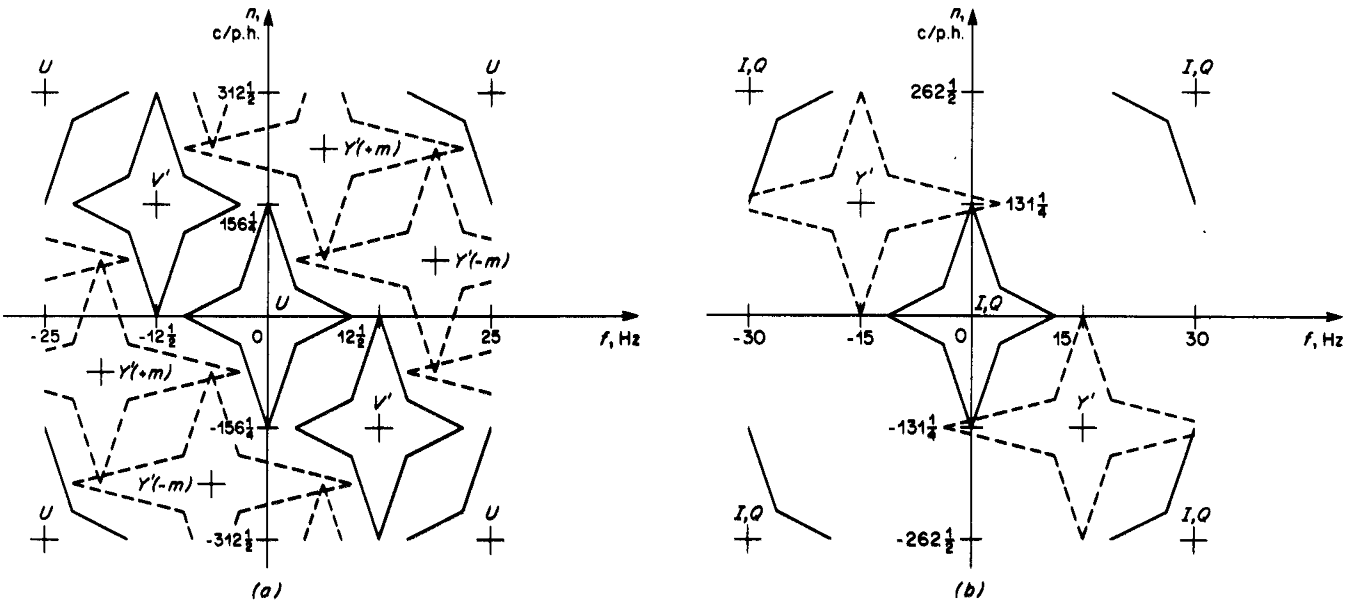


Fig. 60 – Positions of the main chrominance and luminance components in the vertical–temporal spectrum of encoded signals after chrominance demodulation: (a) for 625/50 PAL (U-channel) and (b) for 525/60 NTSC. In (a), Y′ + m and Y′ − m denote the positions of the luminance (cross-colour) components for positive and negative horizontal frequencies, respectively.

will be centred on ($\mp 12\frac{1}{2}$, $\pm 156\frac{1}{4}$) as shown in Fig. 60(a). This accounts for the moving pattern of horizontal lines, known as Hanover bars, which is present in plain coloured areas when a phase distorted PAL signal is decoded using a simple demodulator. In the NTSC system, when phase distortion produces crosstalk between the I and Q signals, this results in a static hue error in plain coloured areas.

4.4. Chrominance filtering

Whereas vertical (comb) filtering in the luminance channel of a PAL decoder is unusual, conventional PAL decoders all use vertical filtering in the two chrominance channels. The filters act to place a zero along the $156\frac{1}{4}$ c/p.h. line of the demodulated signals as shown in Fig. 61 (a), thus removing the V' crosstalk components (Hanover

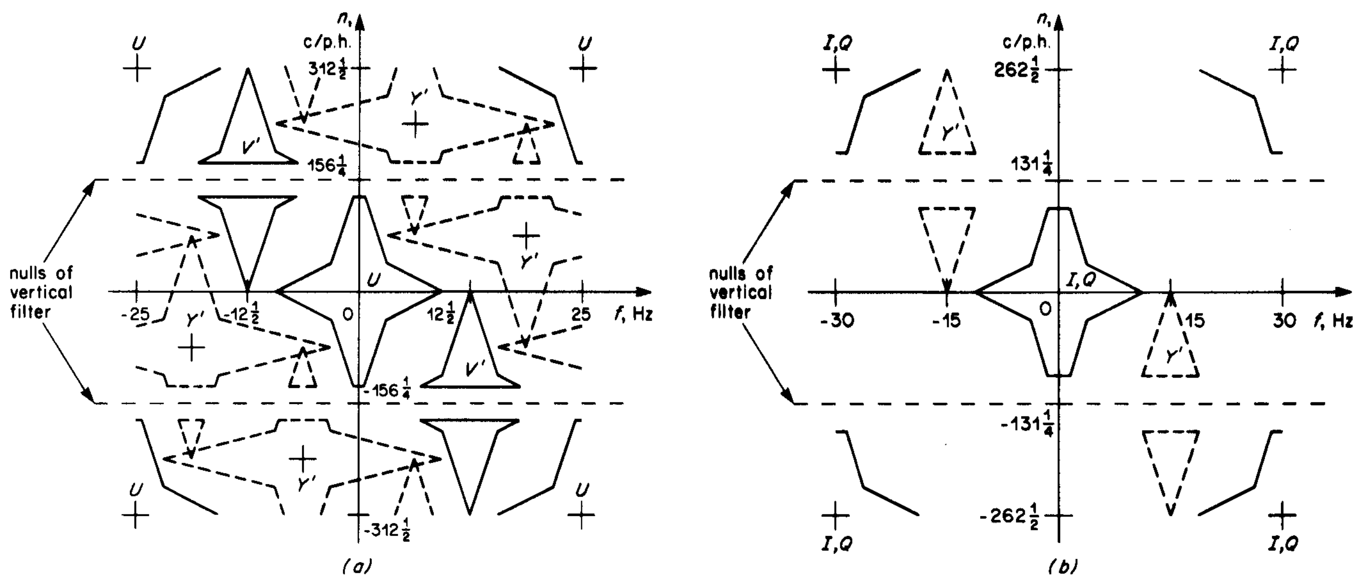


Fig. 61 – The effect of vertical filtering on the spectra of the demodulated chrominance signals shown in Fig. 60: (a) for 625/50 PAL (U-channel) in which the main intention is to suppress co-phased V′ components (Hanover bars) and (b) for 525/60 NTSC in which the intention is to reduce cross-colour.

bars) which would otherwise occur with phase distorted signals. Although the main impairment (Hanover bars in coloured areas) is removed, the vertical sidebands of the crosstalk components remain, Fig. 61 (a). Therefore, on phase distorted signals, errors flashing at $12\frac{1}{2}$ Hz still occur at horizontal transitions between coloured areas. In addition, the vertical filter tends to reduce cross-colour, although the reduction is mostly in the fine, high vertical frequency components. Also, the filtering necessarily introduces a loss of the wanted chrominance components, but this it outweighed by the advantages of reductions in $U-V$ crosstalk and cross-colour. With intra-field (line-based) vertical filters, the frequency characteristics repeat with a period of $312\frac{1}{2}$ c/p.h. so that the filter has unity response at $(0, 312\frac{1}{2})$. This region of extra vertical resolution does not contribute usefully to vertical definition because the loss of lower frequencies is the dominant effect. Instead, the main contribution from this region is cross-colour $(Y' \pm m)$ centred on $(6\frac{1}{4}, 234\frac{3}{8})$.

Some PAL decoders introduce $U-V$ crosstalk as a consequence of vertical filtering.[8] This occurs when contributions from lines of opposite PAL switch sense and incorrect phase reach the demodulators without being shifted to the correct phase. Although these crosstalk components are always vertically filtered so that there is no impairment in areas of constant colour, vertical sidebands of chrominance crosstalk remain, similar to those shown in Fig. 61 (a) at $(\pm 12\frac{1}{2}, \mp 156\frac{1}{4})$.

With the NTSC system, there is no frequency offset between the subcarriers so that the effect of phase errors cannot be reduced by vertical filtering. Nevertheless NTSC decoders often include vertical filters (comb filters based on line delays) purely as a means of reducing cross-colour. The greater separation between the luminance and chrominance signals in the NTSC system allows a broad zero to be placed through the main luminance (cross-colour) positions at $131\frac{1}{4}$ c/p.h. in the demodulated spectrum, Fig. 61 (b), without greatly impairing vertical chrominance resolution. With such a filter, line-repetitive luminance is then completely free of cross-colour.

Being two-dimensional and using contributions from many lines and fields, the interpolator of a standards converter can act to reduce the $U-V$ crosstalk and cross-colour components much more effectively than the line-based filters used in decoders. If a simple demodulator with no vertical filtering is used in the decoder, then the interpolator can reject all the main $U-V$ crosstalk and cross-colour components shown in Fig. 60 (a) without the additional loss of vertical resolution which would occur with a conventional decoder. For example, it is possible to use a frequency characteristic for chrominance interpolation that, for 625/50 inputs, substantially reduces all signals beyond a line joining $(12\frac{1}{2}, 0)$ and $(0, 156\frac{1}{4})$, as shown for the U channel in Fig. 62 (a). This is possible because the colour difference signals are more tolerant of reductions in resolution, particu-

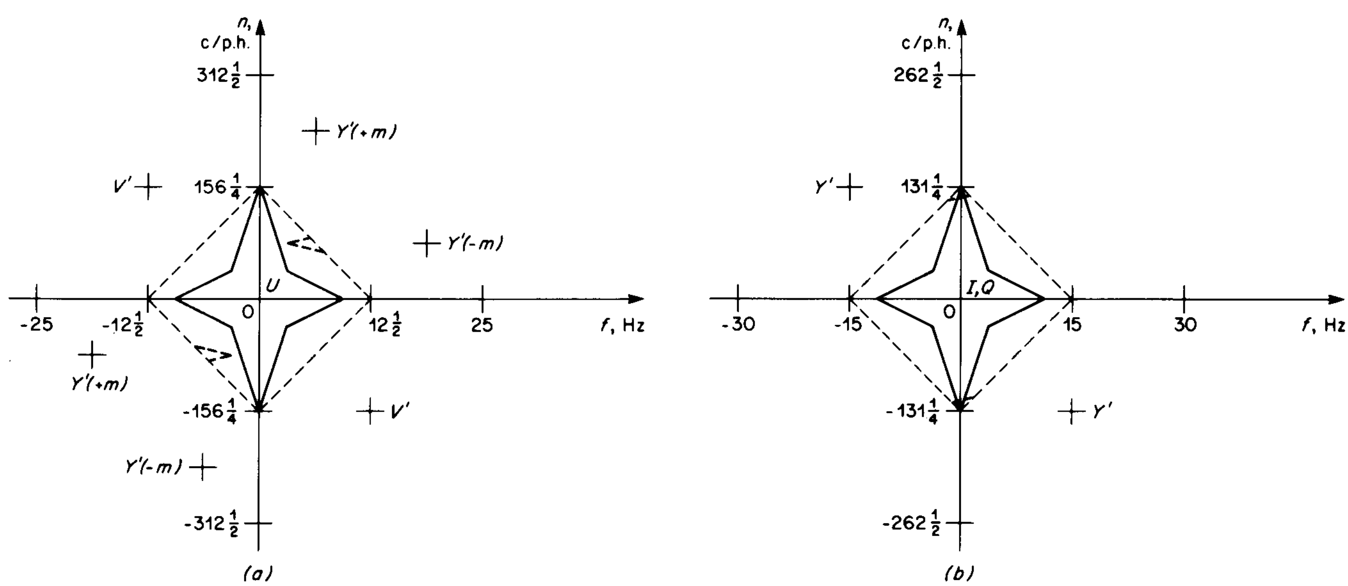


Fig. 62 – Suppression of unwanted components in the demodulated chrominance signals using an interpolation characteristic with reduced vertical and temporal responses: (a) for 625/50 PAL (U-channel) and (b) for 525/60 NTSC. In (a), $Y' + m$ and $Y' - m$ denote the positions of the luminance (cross-colour) components for positive and negative horizontal frequencies, respectively.

larly in the temporal direction, than are luminance signals. This reduction of chrominance resolution has a substantial additional benefit in reducing chrominance noise. A similar approach can be used with NTSC signals, as shown in Fig. 62(b).

The requirements for colour difference signal interpolation are, therefore, substantially different from those for luminance interpolation so it is advantageous to use separate luminance and chrominance interpolation aperture functions. Colour interpolation can then be optimised to obtain the best compromise between the suppression of unwanted components and the retention of adequate vertical and temporal resolution.

5. Practical interpolation apertures

Although the aperture synthesised in Section 3.4 provides the main features required to produce an acceptable frequency characteristic, it can be improved to give better suppression of unwanted scanning products in the two directions of conversion. Also, the procedure presented there does not take account of the need to suppress luminance and chrominance cross-effects and noise. So, the latter part of this Section includes these factors to derive four-field aperture functions suitable for the two directions of conversion and using separate luminance and chrominance interpolation.

However, even with two-field apertures, finer aperture quantisation and combined vertical and temporal interpolation can provide some improvement over the conversion methods described in Section 3.3.2. If this improvement were sufficient, then the extra complexity of a four-field converter would be unnecessary. Furthermore, narrower apertures are of interest to provide fall-back operation in the event of a major failure in part of a four-field interpolator. Therefore, the first part of this Section assesses the performance that can be obtained with two- and three-field apertures.

5.1. Two-field apertures

With contributions from four lines on each field there is no difficulty in obtaining good suppression of vertical aliasing (region C in Fig. 42). However, with only a two-field aperture it is necessary to compromise in temporal performance between the suppression of judder (region D) and the suppression of 5 Hz flicker (region E). The response at 25 Hz can be set to zero to suppress flicker, but this produces a characteristic of sinc*f* shape with large responses between 25 and 50 Hz and beyond 50 Hz. The presence of these responses results in serious movement judder. Alternatively, better suppression of judder can be obtained by using a more gradual temporal characteristic with a broad zero at 50 Hz; but then the 5 Hz flicker components remain in region E. Although flicker components, in general, occur more frequently, judder is a more damaging impairment, particularly at some rates of movement.

Therefore, it is necessary to optimise the temporal characteristic for the best suppression of judder by providing a broad zero at 50 Hz. Flicker can then be reduced by cutting the vertical response along the 25 Hz line.

With separable apertures (which allow vertical and temporal interpolation to be made separate processes) this can only be achieved by cutting all vertical frequencies, including those for still pictures. This results in a frequency characteristic such as that shown in Fig. 63, obtained from the fixed values shown in Table 5.

TABLE 5

vertical frequency (c/p.h.)	0	25	50
$312\frac{1}{2}$	0	0	0
$234\frac{3}{8}$	0	0	0
$156\frac{1}{4}$	0.18	0.0684	0
$78\frac{1}{8}$	0.85	0.3230	0
0	1	0.38	0

temporal frequency (Hz)

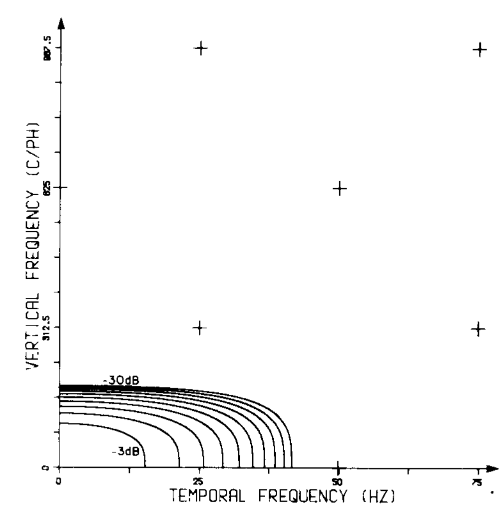


Fig. 63 – The vertical–temporal frequency characteristic of the separable two-field interpolation aperture defined by the fixed-point values of Table 5. Contours are shown at 3 dB intervals down to −30 dB.

This characteristic is broadly similar to that of the DICE converter, Fig. 49, although it includes some small improvements. Finer quantisation of the temporal aperture values has eliminated the response at 75 Hz and reduced the response in the 25 to 50 Hz region, thus reducing judder. The reduced response at 25 Hz also reduces 5 Hz flicker. The slight reduction in vertical resolution compared to Fig. 49 further suppresses 5 Hz flicker, but at the expense of a greater loss of vertical definition.

When vertical and temporal interpolation are combined into one process, the aperture can be made non-separable. Then, for stationary pictures, the vertical resolution obtained with a two-field aperture can be increased, while further reducing the 5 Hz flicker components at 25 Hz. Fig. 64 shows a non-separable frequency characteristic with these features, produced from the fixed point values shown in Table 6. Unfortunately, the inevitable consequence of increasing the vertical resolution and cutting the temporal response more abruptly is the introduction of overshoots in the characteristic, which appear in Fig. 64, centred on ($37\frac{1}{2}, 156\frac{1}{4}$) and ($62\frac{1}{2}, 156\frac{1}{4}$). Although on still pictures the vertical resolution is noticeably greater and the amount of flicker less than for Fig. 63, the movement performance is poorer, being affected by moving high frequency detail from the interlace spectrum centred on ($25, 312\frac{1}{2}$). With two-field apertures, therefore, it is not possible to obtain a good approximation to the desired triangular frequency characteristic. However, the characteristic of Fig.

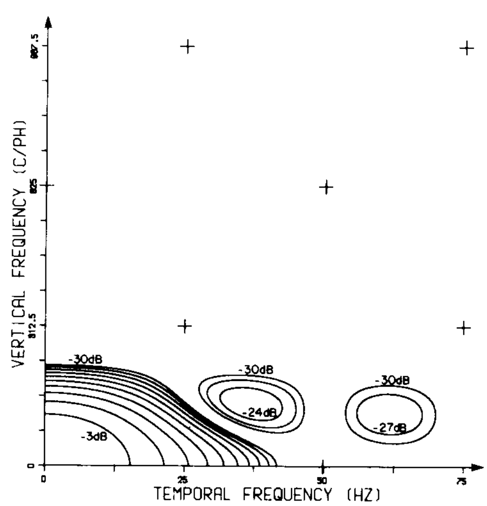


Fig. 64 – The vertical–temporal frequency characteristic of the non-separable two-field interpolation aperture defined by the fixed points of Table 6. The attempt to increase vertical resolution and to reduce 5 Hz flicker has caused significant overshoots in the temporal direction.

63 could be used as a fall-back mode in the event of a component failure in part of a larger converter using more than two fields of storage.

TABLE 6

vertical frequency (c/p.h.)	$312\frac{1}{2}$	0	0	0
	$234\frac{3}{8}$	0.01	0	0
	$156\frac{1}{4}$	0.4	0.01	0
	$78\frac{1}{8}$	0.9	0.22	0
	0	1	0.38	0
		0	25	50
		temporal frequency (Hz)		

5.2. Three-field apertures

Broadening the interpolation aperture to three field periods allows the frequency characteristic to be defined at $16\frac{2}{3}$ Hz intervals. Although a faster rate of cut can be achieved, optimising the design is more difficult because the response values at $16\frac{2}{3}$ and $33\frac{1}{3}$ Hz must be set to obtain the response required in the 25 Hz region. Also, the logic implementation of a three-field converter is generally less straightforward than that for two- or four-field converters, which benefit from being powers of two.

With a three-field converter, making use of the faster rate of cut in the temporal direction produces the characteristic of Fig. 65, calculated from the fixed values shown in Table 7. Whilst maintaining the same vertical resolution as Fig. 64 for still pictures, the three-field characteristic substantially reduces judder and virtually eliminates flicker without producing overshoots in the temporal direction. However, rapid movement is blurred because of insufficient response in the ($12\frac{1}{2}, 0$) region. Extending the characteristic along the temporal axis to reduce blurring could provide a practicable three-field aperture, but there are several clear advantages to be obtained by using four fields instead.

TABLE 7

vertical frequency (c/p.h.)	$312\frac{1}{2}$	0	0	0	0
	$234\frac{3}{8}$	0.01	0	0	0
	$156\frac{1}{4}$	0.4	0.08	0	0
	$78\frac{1}{8}$	0.9	0.3	0	0
	0	1	0.4	0.003	0
		0	$16\frac{2}{3}$	$33\frac{1}{3}$	50
		temporal frequency (Hz)			

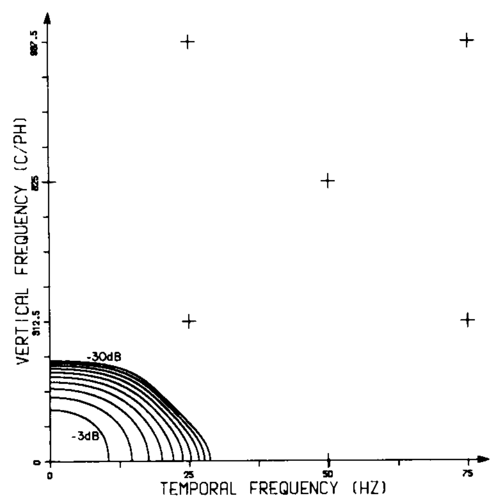


Fig. 65 – The vertical–temporal frequency characteristic of the non-separable three-field interpolation aperture defined by the fixed points of Table 7. With the three-field aperture it is possible to retain the improved vertical response of Fig. 64 without introducing temporal overshoots and to reduce movement judder and 5 Hz flicker.

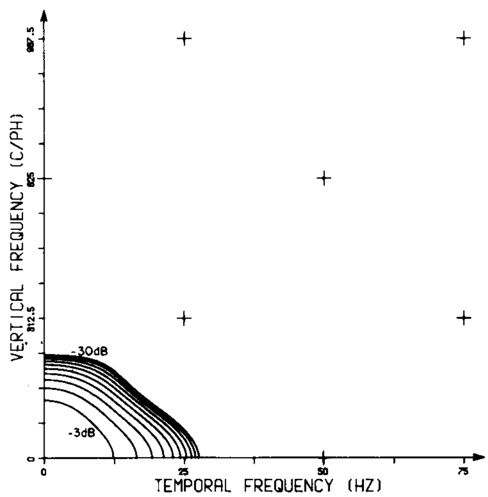


Fig. 66 – The vertical–temporal frequency characteristic of the four-field interpolation aperture for 625/50 to 525/60 conversion defined by the fixed points in Table 8. This four-field aperture provides a close approximation to the ideal triangular shape of Fig. 29. Although judder is well suppressed, rapid movement is too blurred for general use.

5.3. Four-field apertures

Extending the aperture to four field periods allows a faster rate of cut for temporal frequencies. Thus, the characteristic can be well attenuated both at and beyond 25 Hz. With a separable aperture function, this makes the temporal performance more acceptable by reducing judder and flicker with less blurring; but, because the shape of the frequency characteristic is basically rectangular (Section 3.2.1), the vertical resolution cannot be increased. However, with the non-separable case, four-field apertures allow much greater freedom for setting values in the frequency characteristic, so that a closer approximation to the required triangular shape can be obtained. Also, with the finer grid of points, shown in Fig. 41 (a), it is possible to develop different characteristics for luminance in the two directions of conversion and to provide a separate aperture for chrominance interpolation.

5.3.1. 625/50 to 525/60 luminance conversion

When converting from 625 to 525 lines, the output can accommodate less vertical resolution than the input standard. Therefore, in order to avoid interlace flicker on the output picture, it is necessary to suppress frequencies of $262\frac{1}{2}$ c/p.h. and above. Frequencies below this limit should be retained as far as possible so that reasonable picture sharpness is maintained.

In contrast, with a change in field rate from 50 to 60 Hz, the output standard has the capacity for higher temporal frequencies. Even so, frequencies of 25 Hz and above must be well attenuated to avoid 5 Hz flicker and judder. These requirements combine to produce the frequency characteristic of Fig. 66, obtained from the fixed values of Table 8.

TABLE 8

vertical frequency (c/p.h.)					
$312\frac{1}{2}$	0	0	0	0	0
$234\frac{3}{8}$	0.02	0	0	0	0
$156\frac{1}{4}$	0.5	0.125	0	0	0
$78\frac{1}{8}$	0.95	0.49	0.01	0	0
0	1	0.7	0.1	0	0
temporal frequency (Hz)	0	$12\frac{1}{2}$	25	$37\frac{1}{2}$	50

Compared with the three-field characteristic, Fig. 65, this four-field aperture retains slightly more resolution along the vertical axis and cuts more sharply to suppress frequencies above $262\frac{1}{2}$ c/p.h. This is only possible because the temporal cut-off can be more abrupt, so that more vertical resolution is provided for stationary pictures, while the alias components resulting from moving high vertical frequency components are still suppressed. In addition, the on-axis temporal characteristic has a slightly sharper cut and the triangular shape is more pronounced, resulting in a characteristic much closer to the ideal than could be obtained with three fields.

In practice, the performance on still and slowly moving pictures is near ideal, being free from flicker and other impairments, and providing good spatial resolution. However, rapid movement is blurred due to inadequate response in the $12\frac{1}{2}$ to 25 Hz region. The blurring can be reduced by extending the characteristic to give more response along the temporal frequency axis. Fig. 67 shows a characteristic with the same vertical response but with an extended temporal response, obtained from the fixed point values shown in Table 9. This characteristic greatly reduces blurring on rapid movement, but leaves a much higher level of judder, which occurs at relatively slow rates of movement from high spatial frequencies. Components along the 25 Hz line, which cause 5 Hz flicker, remain at an acceptably low level.

TABLE 9

vertical frequency (c/p.h.)						
	$312\frac{1}{2}$	0	0	0	0	0
	$234\frac{3}{8}$	0.02	0	0	0	0
	$156\frac{1}{4}$	0.5	0.15	0	0	0
	$78\frac{1}{8}$	0.95	0.6	0.07	0	0
	0	1	0.9	0.32	0	0
temporal frequency (Hz)		0	$12\frac{1}{2}$	25	$37\frac{1}{2}$	50

It is necessary, therefore, to distinguish between two distinct types of movement. First, there are the rapid, short duration movements which occur within a scene, such as the arm and leg movements of a dancer. This type of movement predominates in studio pictures. Because each movement is of short duration, there is no opportunity for the viewer to track the movement, so that judder, if present, is not noticeable. However, blurring of these rapid movements is noticeable and can cause objectionable impairments if the temporal characteristic cuts at too low a frequency. Therefore, for studio pictures, the characteristic of Fig. 67 is best.

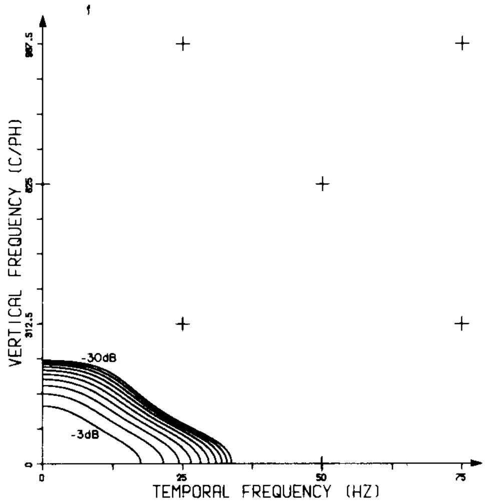


Fig. 67 – The vertical–temporal frequency characteristic of the four-field aperture for 625/50 to 525/60 conversion defined by the fixed-point values in Table 9. The extended temporal response avoids blurring but leaves a higher level of judder than occurs with the characteristic of Fig. 66. This extended characteristic represents the best overall compromise for suppressing judder and avoiding blurring.

The second type of movement is relatively slow, steady and of long duration, such as occurs in the background when a camera is panning to follow a moving foreground object. This camera-induced movement frequently occurs during outside broadcasts, particularly at sporting events. As the original movement is steady and of long duration, the viewer is able to recognise and follow the natural rate of movement. Then, the unnatural movement judder perturbations, due to aliasing, become more obvious. With this type of movement, although blurring of the background is noticeable, it is a much more acceptable impairment. Therefore, for outside broadcasts, the frequency characteristic of Fig. 66 offers some advantages.

This inability to provide satisfactory performance under all conditions of movement is a direct result of the inadequacy of the pre-sampling low-pass filter (camera integration) and undersampling (using too low a field rate). Because of these factors, the wanted movement and unwanted alias components in the television signal share the same spectrum space and cannot be distinguished. Therefore, blurring of true movement is a necessary consequence of suppressing judder and, if true movement is not blurred, judder will also be present. Choosing a characteristic between those of Figs. 66 and 67 would result in neither type of movement being particularly well portrayed. The use of a more abrupt temporal cut-off, which would require more field stores, could provide better retention of the main wanted components and better suppression of the main unwanted components. However, as the true and alias components appear to overlap extensively, it is possible that a sharper filter might give no improvement at all. An alternative approach could be to use the most suitable aperture (corresponding to either Fig. 66 or Fig. 67) for each picture, under the control of a panning detection signal. While this might lead to some improvement, the abrupt change of mode itself might be found disturbing. In addition, the inclusion of such a system could result in a substantial increase in interpolator complexity.

Without this refinement, the characteristic of Fig. 67 is generally preferred, particularly by

untrained observers.* This is because, although judder is usually a more objectionable impairment than blurring, it can go unnoticed by a viewer whose attention is following a foreground object. However, viewers whose attention is caught by excessive background judder are subsequently more likely to notice judder at lower levels. This learning process suggests that viewers will gradually find judder less acceptable.

With regard to cross-luminance, the two characteristics, especially that of Fig. 66, give a useful degree of suppression of the main PAL subcarrier components, centred on $(6\frac{1}{4}, 234\frac{3}{8})$ and $(18\frac{3}{4}, 78\frac{1}{8})$. Nevertheless, the main consideration in selecting a luminance frequency characteristic is the suppression of scanning products and this precludes additional optimisation of cross-luminance suppression.

5.3.2. 525/60 to 625/50 luminance conversion

While many of the principles discussed in the previous Section also apply for 525/60 to 625/50 conversion, the spacings of the frequency specification points change to 15 Hz for temporal frequency and $65\frac{5}{8}$ c/p.h., for vertical frequency. Because of this, the temporal characteristic has to cut more rapidly to take account of the lower temporal resolution of the output standard. Also, it would be advantageous to extend the response along the vertical frequency axis to take advantage of the higher vertical resolution of the output standard. However, because of the limited rate of cut in the temporal direction, only minor changes are possible without either losing the triangular shape of the characteristic or introducing ringing in the temporal direction. Incorporating these considerations produces the characteristic of Fig. 68, calculated from the fixed point values shown in Table 10.

*Those used to normal picture impairments, but unfamiliar with impairments specific to standards conversion.

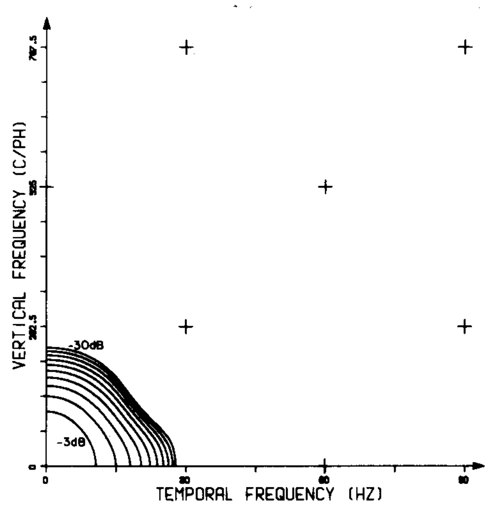


Fig. 68 – The frequency characteristic of a four-field aperture for 525/60 to 625/50 conversion defined by the fixed points in Table 10. As with Fig. 66, rapid movement is too blurred for general use.

TABLE 10

vertical frequency (c/p.h.)						
	$262\frac{1}{2}$	0	0	0	0	0
	$196\frac{7}{8}$	0.1	0	0	0	0
	$131\frac{1}{4}$	0.5	0.11	0	0	0
	$65\frac{5}{8}$	0.9	0.35	0	0	0
	0	1	0.5	0	0	0
temporal frequency (Hz)		0	15	30	45	60

Although the vertical resolution is rather limited, this characteristic gives output pictures free from 5 Hz flicker and vertical aliasing. As with Fig. 66, movement is free of judder, but the more rapid movements are blurred. Again, blurring can be avoided, at the cost of reintroducing judder, by extending the temporal frequency characteristic as shown in Fig. 69, which corresponds to the fixed values of Table 11. As before, this extended characteristic is to be preferred for general use, although the characteristic of Fig. 68 produces more acceptable pictures during camera panning.

TABLE 11

vertical frequency (c/p.h.)						
	$262\frac{1}{2}$	0	0	0	0	0
	$196\frac{7}{8}$	0.1	0	0	0	0
	$131\frac{1}{4}$	0.5	0.15	0	0	0
	$65\frac{5}{8}$	0.9	0.5	0.02	0	0
	0	1	0.75	0.15	0	0
temporal frequency (Hz)		0	15	30	45	60

The NTSC subcarrier, located at $(15, -131\frac{1}{4})$, is well attenuated, particularly with the characteristic of Fig. 68. However, horizontal colour transitions, which produce high vertical modulating frequencies, result in components falling near the temporal axis and these receive much less attenuation. After comb filtering, much of the residual subcarrier energy falls in this region. Furthermore, for moving vertical chrominance detail, the attenuation can fall to zero and, under these circumstances, the cross-luminance pattern of the NTSC signal becomes visible on the 625/50 output standard pictures.

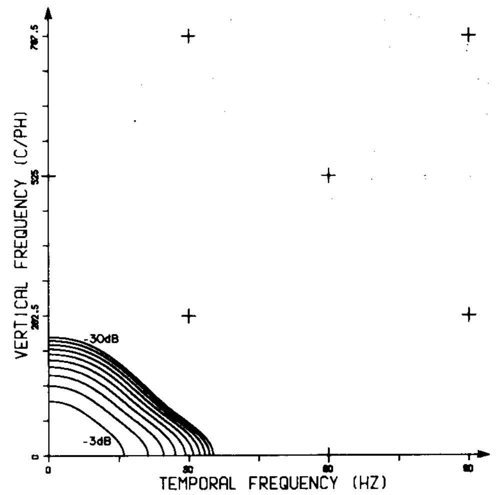


Fig. 69 – The frequency characteristic of a four-field aperture for 525/60 to 625/50 conversion defined by the fixed points in Table 11. As with Fig. 67, blurring is avoided, but there is more judder than with Fig. 68.

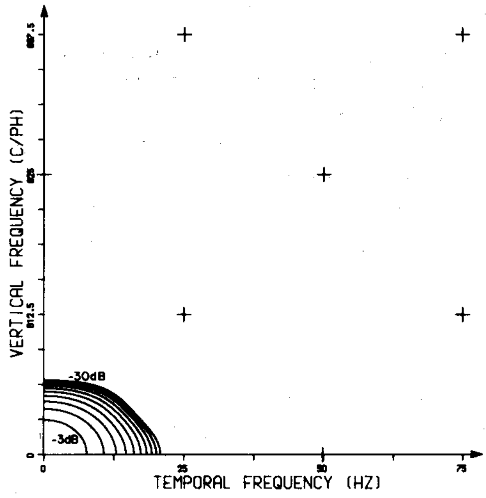


Fig. 70 – The frequency characteristic of a four-field aperture for chrominance interpolation defined by the fixed-point values in Table 12. The temporal response is rapidly curtailed, whilst the vertical response gives negligible softening of coloured vertical detail.

5.3.3. Chrominance conversion

With chrominance signals, the retention of vertical and temporal resolution is much less important than for luminance signals. Therefore, it is advantageous to sacrifice resolution in order to suppress cross-colour, Hanover bars and noise. Because high temporal frequencies are virtually absent from true colour difference signals (the result of their low spatial resolution), a very rapid temporal cut-off can be used without noticeably blurring the signals. Also, in the vertical direction, a cut-off similar to that given by a delay line decoder results in a barely perceptible loss of resolution. Placing a zero at $(12\frac{1}{2}, 156\frac{1}{4})$ ensures complete suppression of Hanover bars in areas of constant colour. This produces the characteristic of Fig. 70, which is obtained from the fixed values shown in Table 12.

TABLE 12

vertical frequency (c/p.h.)	$312\frac{1}{2}$	0	0	0	0	0
	$234\frac{3}{8}$	0	0	0	0	0
	$156\frac{1}{4}$	0.075	0	0	0	0
	$78\frac{1}{8}$	0.7	0.2	0	0	0
	0	1	0.38	0	0	0
		0	$12\frac{1}{2}$	25	$37\frac{1}{2}$	50

temporal frequency (Hz)

As well as suppressing Hanover bars, this characteristic produces pictures with substantially reduced cross-colour and chrominance noise, while introducing virtually no impairment to the wanted chrominance components. However, an alternative characteristic, Fig. 71, calculated from the values of Table 13, produces greater suppression of cross-colour and noise by further sacrificing vertical resolution. Although noticeable, this loss of resolution is not serious, while the additional suppression of noise has a beneficial effect on very noisy signals from poor quality sources.

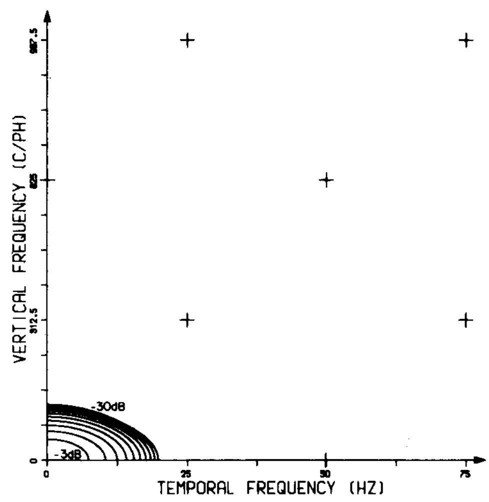


Fig. 71 – The frequency characteristic of a four-field aperture for chrominance interpolation defined by the fixed-point values in Table 13. Compared with Fig. 70, the vertical response has been further reduced to provide increased suppression of chrominance noise and cross-colour.

TABLE 13

vertical frequency (c/p.h.)					
312½	0	0	0	0	0
234⅜	0	0	0	0	0
156¼	0	0	0	0	0
78⅛	0.35	0.07	0	0	0
0	1	0.35	0	0	0
	0	12½	25	37½	50

The same aperture functions can also be used for 525/60 to 625/50 chrominance conversion and then scaled versions of the frequency characteristics are produced. Again, the resulting converted pictures contain substantially less cross-colour than unconverted pictures and chrominance noise components are greatly reduced.

With the experimental converter, it was possible to demonstrate the action of an interpolation method directly in the frequency domain using a test pattern generator.[12] Fig. 72 (a) shows a two-dimensional sweep pattern produced by the generator in which vertical frequencies increase in the vertical direction and temporal frequencies increase in the horizontal direction from a central origin. The centre of the screen therefore corresponds to (0, 0) and the corners to (± 25, $\pm 312\frac{1}{2}$) in the *n–f* spectrum. The repeated patterns at the corners of the screen are the normal result of interlaced scanning. Fig. 72 (b) shows the result of converting this signal using a low resolution interpolation method similar to that shown in Fig. 71. The filtering effect of the conversion process, particularly for vertical frequencies, is readily apparent.

6. Conclusions

Television scanning is a sampling process which repeats the vertical and temporal frequency components of an image at harmonics of the scanning rates. Converting from one scanning standard to another therefore consists of changing the sampling frequencies of the scanning process.

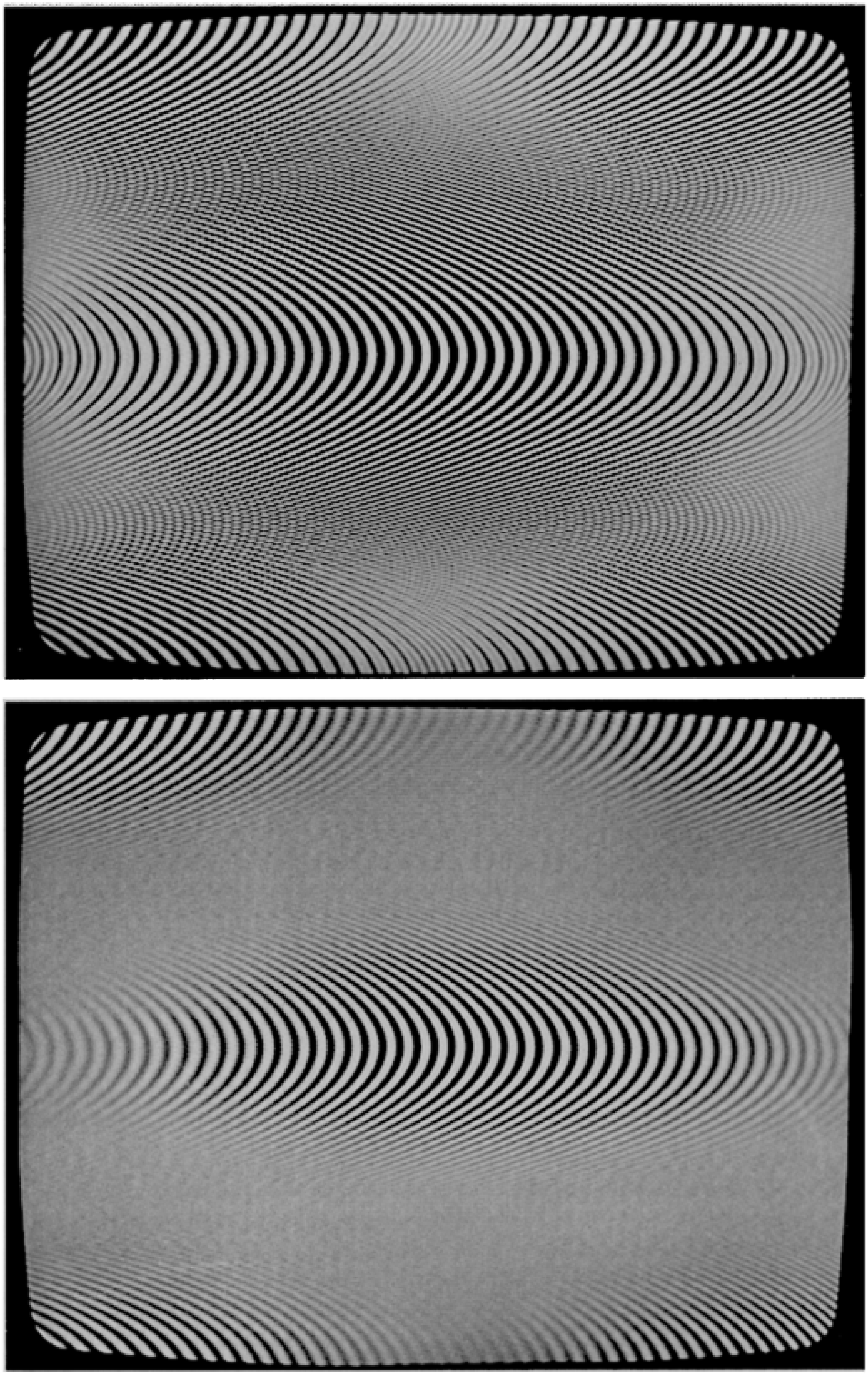
In a digital standards converter, signal information is transferred from the lines and fields of the input standard to those of the output standard by interpolation – a process equivalent to a combination of low-pass filtering and resampling. In practice, it is instructive to visualise interpolation in these terms, so that performance depends on the effectiveness of the equivalent low-pass filter in suppressing the harmonic spectra introduced by the first scanning process whilst retaining the original baseband spectrum of the image. The interfering effect of the residual harmonic spectra that occurs with resampling on the output standard can then be clearly seen. To illustrate this, the theory of sampling rate conversion has been developed in general terms for the one-dimensional case. Of particular importance is a method of interpolation aperture synthesis by which an aperture of specified width can be produced from values fixed in the low-pass filter frequency characteristic.

There are no limits to the vertical and temporal frequencies which can occur in a scene. Although the television camera acts as a presampling low-pass filter, this, in the temporal dimension at least, is insufficient to avoid overlap between the spectral components. The resulting aliasing is an accepted feature of normal television pictures, but after standards conversion the alias components can become more noticeable. Because these 'in-built' alias components are present, perfect standards conversion is impossible, even in theory.

With interlaced scanning, the repeated spectra are arranged in an offset structure. In order to reject these offset components, the two-dimensional frequency characteristic of the interpolation process requires a triangular shape. If vertical–temporal interpolation is carried out as two one-dimensional processes, the resulting two-dimensional characteristic is rectangular. With that form of interpolation, it is necessary to sacrifice vertical resolution in order to obtain reasonable performance in other respects. It is preferable, therefore, to combine vertical and temporal interpolation into a single two-dimensional process. Although this results in an increase in complexity, the required triangular shape can be obtained and the loss of vertical resolution avoided.

With signals derived from composite PAL or NTSC, the normal low-pass filtering action of the vertical–temporal interpolator tends to suppress the main cross-luminance and cross-colour components left by the decoding process. In the colour difference signals, however, resolution, particularly temporal resolution, is much less important than for luminance. Therefore, by using different interpolation apertures for luminance and chrominance, it is possible to sacrifice chrominance resolution to obtain much better suppression of cross-colour, Hanover bars and chrominance noise.

Using a point fixing method described in Section 3.4, different aperture functions have been optimised for converters using two, three and four fields of storage. With a two-field aperture, even with the greater flexibility of combined vertical and temporal interpolation, it proved impossible to suppress both judder and 5 Hz flicker impairments,



◁ (a)

◁ (b)

Fig. 72 — Display of the vertical–temporal frequency response of an interpolation method: (a) An electronically generated test pattern producing a sweep of vertical frequencies vertically and temporal frequencies horizontally on the 625/50 standard. The centre of the screen corresponds to (0,0) and the corners to (±25, ±312½). (b) The same signal converted to the 525/60 standard using an aperture function suitable for chrominance interpolation (frequency characteristic similar to that shown in Fig. 71) in the experimental standards converter.

and vertical resolution was restricted. Although reasonable performance could be obtained with a three-field aperture, worthwhile benefits resulted from using four fields. Thus it was possible to eliminate all flicker and aliasing effects on still pictures while maintaining high resolution. Impairments to movement remained, however, although these were substantially less than with previous converters.

Two distinct types of movement were identified which required different treatment: rapid movements within a scene and the rather slower, uniform movements which result from camera panning. When optimising the interpolation for rapid movement, it was found that little blurring could be tolerated so requiring a wide temporal characteristic. However, for camera panning, the level of judder produced with such a wide characteristic was very noticeable. Conversely, when the interpolation was optimised to reduce judder on panning, the blurring of rapid, short duration movements was unacceptable. Overall, the wider temporal characteristic was found to be more acceptable.

To extend the temporal aperture to use more than four fields would allow a more rapid rate of cut for temporal filtering. However, this would not necessarily provide any additional improvement because the repeated spectral components resulting from scanning, overlap with the baseband spectrum in the temporal direction.

With a separate four-field aperture producing a relatively narrow frequency characteristic for chrominance, it was possible to eliminate Hanover bars and to reduce cross-colour significantly, while maintaining more spatial resolution than could be obtained with a normal PAL or NTSC decoder. This narrow chrominance characteristic also noticeably reduced chrominance noise. In these respects, therefore, converted pictures produced by a four-field interpolator were superior to the same pictures conventionally decoded on the input standard.

Four-field apertures of the types described here are used in the two production standards converters, known as ACE*, made for the Television Service by BBC Engineering Designs Department.

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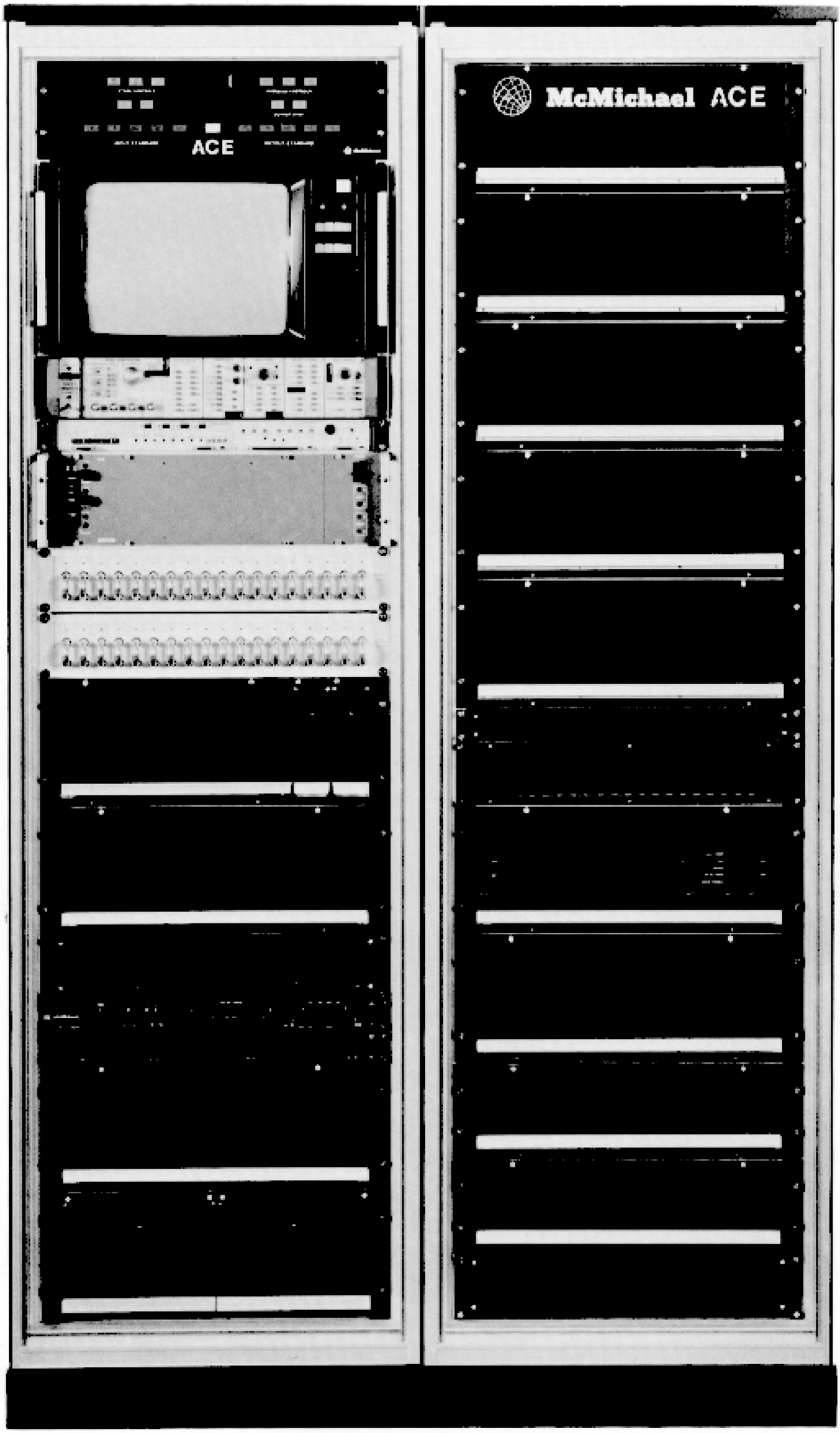
*ACE – Advanced Conversion Equipment. ACE is being manufactured for general sale by GEC – McMichael Ltd.



The GEC-McMichael ACE standards converter based upon research, development and design work done in Research and Designs Departments.